

LINK-ADAPTIVE MEDIUM ACCESS CONTROL PROTOCOLS FOR HIGH-SPEED
WIRELESS NETWORKS

By

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A DISSERTATION PRESENTED TO THE GRADUATE SCHOOL
OF THE UNIVERSITY OF FLORIDA IN PARTIAL FULFILLMENT
OF THE REQUIREMENTS FOR THE DEGREE OF
DOCTOR OF PHILOSOPHY

UNIVERSITY OF FLORIDA

2004

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This dissertation is dedicated to my beloved family, who supported me emotionally and financially throughout this long journey.

ACKNOWLEDGMENTS

My dissertation could not have been completed without the help and support of certain people. I could not possibly list all those to whom I owe my gratitude, but I would like to mention some of the most important names.

I am deeply indebted to my supervisory committee chair, Professor Yuguang Fang. His stimulating discussions and encouragement helped me to carry out the research in this dissertation. Despite his busy schedule, he always set aside plenty of time to discuss even small ideas with me; and to offer guidance, and patiently monitor my research.

I would also like to thank my cochair, Professor Tan Wong, for his immeasurable contributions to my research and for expanding my research motivation to a physical-layer in wireless communication.

I thank Professor John Shea and Professor Shigang Chen, who provided me with insightful suggestions that greatly improved the quality of this dissertation. In particular, Professor John Shea advised me on my curiosities about the wireless world from the beginning of my Ph.D. study.

My wonderful colleagues in the Wireless Network Laboratory (WINET) are like brothers and sisters; they have supported and inspired me throughout my research work. I express my sincere appreciation for their help, support, interest, and invaluable hints. I will never forget the night when all of us went to meet Professor Yuguang Fang at the airport. I felt a true kinship.

Finally, I would like to express my special appreciation to my parents, Bok-Soo Kim and Ok-Ja Chung, who provided the ideal environment for me to grow up. They have encouraged my interest in sciences. I give heartfelt thanks my lovely wife, Jung Suk Lee, whose understanding and constant support made this entire endeavor worthwhile; and my son, Sung-Hyun Kim, who has given me many delights in daily life.

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Abstract of Dissertation Presented to the Graduate School
of the University of Florida in Partial Fulfillment of the
Requirements for the Degree of Doctor of Philosophy

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December 2004

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Medium Access Control (MAC) protocols for wireless networks have been studied extensively to support broadband communication services with various Quality-of-Service (QoS) requirements. Our study aimed to improve the performance of MAC protocols using channel information in high-speed wireless networks such as wireless local area networks (WLANs) and wireless personal area networks (WPANs).

First, a rate-adaptive protocol with dynamic fragmentation was proposed to enhance the throughput based on fragment transmission bursts and channel information. Instead of using a fragmentation threshold as in the IEEE 802.11 standard, I introduced multiple thresholds for different data rates, so that more data could be transmitted at higher data rates when the channel is good. In the proposed scheme, the channel can be used more effectively to squeeze more bits into the medium.

Second, I proposed an efficient polling-based MAC protocol, referred to as Two-Step MultiPolling (TS-MP), with the goal of serving as a centralized polling-based

channel access method that also supports time-bounded services. The TS-MP protocol uses two multi-polling frames for different purposes. The first polling frame is broadcast to collect information such as the numbers of pending frames and the physical-layer transmission rates for the communication links among all stations, which help to implement rate adaptation over the time-varying wireless channel. The second polling frame contains a polling sequence for data transmissions that was designed based on the collected information.

Finally, I proposed a feedback-assisted and link-adaptable MAC protocol and an efficient channel-time allocation algorithm for delay-constrained real-time traffic in WPANs. Channel time for each node is initially allocated based on statistical packet inter-arrival time. Then, the initial allocation is dynamically adjusted by using feedback information coming from each DEV. Feedback information includes buffer status, packet transmission delay, and physical transmission rate. From the buffer status and rate information, the central DEV can allocate sufficient channel time for transmissions of pending packets at a DEV. In addition, the allocated channel times can be synchronized to the packet arrival time using the feedback information. This reduces the overall transmission delay. To cope with time-varying wireless channels, a dynamic rate-selection algorithm assisted by physical-layer information is proposed.

CHAPTER 1 INTRODUCTION

Ubiquitous access to information has long been a dream for mankind. The wireless channel is the only communication medium that can enable anywhere, anytime, tetherless communication. With the recent advances in wireless technologies, it is now possible to build high-speed wireless systems that are cheap as well as easy to deploy and use. Mobile and portable telephone and data services have been influencing our daily lives for some years now. Moreover, widespread deployments of wireless networks have revolutionized communications and information processing in business and private applications.

Using the wireless channel as a communications medium can enable multiple devices to access the medium at the same time, so that multiple simultaneous transmissions are possible. However, the transmission quality may suffer deterioration, since simultaneously transmitted signals cause interference with each other's receivers. In an effort to solve this problem, wireless Medium Access Control (MAC) protocols have been studied extensively since the 1970s. These MAC protocols define rules that allow numerous communication devices to communicate with each other in an orderly and efficient manner. Consequently, MAC protocols play a crucial role in enabling multiple accesses by ensuring efficient and fair sharing of the scarce wireless bandwidth. MAC protocols were initially developed for data and satellite communications [1-2]. Because of the convergence of voice, data, and video applications in wireless communication networks, quality of service (QoS) requirements for real-time traffic in wireless networks

has recently become another important issue in wireless network MAC research.

Consideration of the QoS requirement has led to novel and complex MAC protocol developments.

In the past quarter-century, we have witnessed the rollout of three generations of wireless cellular systems providing efficient mobile communications to end-users. On another front, wireless technology has also become an important component in providing networking infrastructure for localized data delivery at higher speeds. This later revolution was made possible by the introduction of new networking technologies and paradigms such as wireless local area networks (WLANs) and wireless personal area networks (WPANs).

The best known WLAN (IEEE 802.11 WLAN) was designed to support portable computing devices using broadband wireless access in businesses and homes [3]. As broadband technology has become more widely available and demand for the next level of broadband functionality accelerates, WLAN has emerged as the leading technology to satisfy this demand. Furthermore, WLAN is viewed as the edge network of choice for the futuristic 4G cellular network. As a consequence, the IEEE 802.11 standard has rapidly evolved from the 802.11 Task Group (TG) a to TG n. Starting from the 2 Mbps data rate in the physical-layer, it has evolved to the 54 Mbps data rate. The IEEE 802.11n TG seeks to achieve an even higher data rate (at least 100 Mbps) in the near future [3].

The MAC protocol in IEEE 802.11 [4] consists of two coordination functions: Distributed Coordination Function (DCF) and Point Coordination Function (PCF). These two functions define the structure of WLAN as a distributed or centralized network. In the DCF, a set of wireless stations (STAs) communicates directly with each other without

any coordination from a centralized controller, by using a contention-based channel access method, Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA). In the PCF, the channel access of each station is controlled by polling from a Point Coordinator (PC) at the Access Point (AC). While the DCF is designed for asynchronous data transmission, the PCF is mainly intended to support time-bounded services such as voice and video. The DCF and PCF can coexist by alternating Contention Free Periods (CFPs) ruled by the PCF and Contention Periods (CPs) ruled by the DCF.

Since the IEEE 802.11 MAC protocol was standardized without considering of any significant QoS support, enhancements to the IEEE 802.11 MAC are an impediment in deploying multimedia applications. Therefore, IEEE 802.11 TGe recently proposed an enhanced function, called the Hybrid Coordination Function (HCF) [5], to support the QoS required services efficiently. In the HCF, the AC is allowed to start a CFP at any time during a CP, and the channel access in the CFP is controlled by the polling method in IEEE 802.11.

The Wireless Personal Area Networks (WPANs) being studied by the IEEE 802.15 Working Group (WG) enable short-range wireless connectivity among consumer electronics and communication devices with low transmission power. This is in contrast to WLANs, which usually provide a larger transmission range. The transmission range of WPAN is around 5 to 50 meters [6, 7]. The IEEE 802.15 WG has evolved into five TGs, according to the target applications or tasks.

The IEEE 802.15 WG TGs 3 and 4 explore the adaptation of Ultra Wide Band (UWB) technology as the physical-layer technologies for applications in WPAN and wireless sensor networks, respectively. The IEEE 802.15.3 standard [8], named High-

Rate (HR) WPAN, is aimed at consumer electronics and portable communication devices requiring higher data rates. HR WPAN's target applications are multi-megabyte data file transfers such as image and music files and distribution of real-time video and high-quality audio. The HR WPAN standard supports data rates of up to 55 Mbps. Since the Federal Communications Commission (FCC) approved the commercial use of UWB technology [9], the IEEE 802.15.3a Study Group (SG) has been established to study UWB technology as a physical (PHY) layer transmission technique in HR WPAN. Using UWB technology, the maximum achievable data rate can exceed 500 Mbps [6]. On the other hand, the IEEE 802.15 TG 4 is chartered to investigate a low-data-rate solution with a multi-month to multi-year battery life and very low complexity. It is intended to operate in an unlicensed, international frequency band. Potential applications include sensors, interactive toys, smart badges, remote controls, and home automation.

The MAC layer specifications in IEEE 802.15 TGs 3 and 4 are designed to support ad hoc networking, where a node can have either master or slave functionality based on the existing network conditions. Every node can easily join or leave an existing network. The nodes communicate on a centralized and connection-oriented ad hoc networking topology. Although the MAC specifications of both TGs operate under the same superframe structure, a pair of nodes communicates mainly without contention during the channel time allocated by a scheduler in a piconet controller (PNC) in the IEEE 802.15.3 standard [8]. On the other hand, contention-based channel access is used in the IEEE 802.15.4 standard [10].

Although the standards have evolved in the course of adopting newly advanced technologies to meet the demands of consumers and industries, the standardized MAC

protocols have been evaluated extensively and have been enhanced in aspects such as QoS guarantees, overheads, fairness, link adaptation, throughput, and power conservation [11-13]. Moreover, the standard MAC protocols leave some parts unspecified. In this regard, a plethora of proposals have been suggested to enhance the protocols and to address the unspecified aspects of the standards in the current literature [14-44].

Traditionally, protocols at different layers are designed separately in order to achieve modularity and portability. However, integration and coordination across layers have received intensive attention [16-18, 20-29, 45-56]. Joint design of the MAC and physical-layers has been investigated [16-18, 20-29, 53-56]. In the physical-layer specifications of the aforementioned standards, multiple data rates are supported by using variable modulators and encoders. Reliable and high-bit-rate transmissions can be achieved based on the established communication link. Since the wireless link is time-varying in nature, it is desirable that the physical transmission data rate dynamically changes according to the link condition. Because the wireless channel may be asymmetric, the transmitter needs to obtain a feedback frame containing the channel information from the receiver before transmitting a data packet. This mechanism can be implemented by using the MAC protocol. However, the MAC protocol must be matched to the physical-layer for better resource use. Knowing the link condition reported to the MAC layer by the physical-layer helps to achieve higher performance in wireless communication systems.

This dissertation addresses the designing of efficient MAC protocols adaptable to time-varying wireless communication links.

CHAPTER 2 DYNAMIC FRAGMENTATION SCHEME IN WLANS

2.1 Introduction

A typical wireless communication link is time-varying, so it is challenging to design transmission schemes more effectively based on the channel condition. Many adaptive transmission schemes for enhancing throughput performance have been proposed in the literature. Many of these schemes vary the data rate, transmission power, or packet length. One of the popular schemes is based on rate adaptation. This scheme includes an adaptive transmission method that uses different modulation and coding schemes to adjust the data rate based on the channel condition in terms of the Signal-to-Noise Ratio (SNR). The basic idea is to use a higher-level modulation scheme when a higher SNR is detected, as long as the target error rate is satisfied. The target error rate can be characterized by the Bit Error Rate (BER), the Symbol Error Rate (SER), or the Packet Error Rate (PER), as specified by the designer. For receiver-based rate-adaptation schemes, the receiver usually carries out the channel estimation and rate-selection, and the selected rate is then fed back to the transmitter.

Many rate-adaptation schemes for 2.5G and 3G wireless cellular networks using centralized TDMA-based MAC protocols have been proposed [16, 53-56]. Power control schemes based on power conservation and rate adaptation have also been proposed [17, 18]. All of these schemes work at the base station in a centralized fashion. However, to our knowledge, only a few MAC protocols with rate adaptation have been proposed for distributed wireless local area networks (LANs). The Auto Rate Fallback (ARF) protocol

[19] is proposed. In the ARF, the sender selects the rate based on the packet transmission failure rate. Whenever transmission failures occur, a lower rate is chosen. Performance of this scheme with threshold selection for fallback was evaluated [21]. A similar scheme [20] was designed based on the timeout of the acknowledgement (ACK) frame. The RTS/CTS collision avoidance handshaking was used [21-23] to exchange channel information, and then to select the rate accordingly. Specifically, in the Receiver-Based AutoRate (RBAR) MAC protocol proposed [21], channel estimation and rate selection are carried out by the receiver based on the RTS transmission; and the selected rate is sent back to the sender in the MAC header of the CTS packet. To enhance transmission reliability of the MAC header, a cyclic redundancy check (CRC) code is added to the MAC header in RBAR. A two-step scheme was proposed to update the Network Allocation Vector (NAV), which is a critical component in the virtual carrier sensing in IEEE 802.11 MAC. More details on this scheme are given in Section 2.2. It has been shown that RBAR achieves better throughput performance than ARF does, because rate adaptation based on channel estimation can better cope with the time-varying nature of the channel.

With ARF and RBAR, the sizes of transmitted packets vary; hence, all nodes may have different channel access times. This may aggravate the unfairness issue in time. Therefore, the Opportunistic Auto Rate (OAR) protocol proposed [23] suggests allowing a sender to use a high data rate to transmit more packets for the duration of the time for which the sender has acquired the channel access right.

In the IEEE 802.11 standard, when a MAC Service Data Unit (MSDU) generated by the Logical Link Control (LLC) layer is larger than the fragmentation threshold, the

MSDU is fragmented into smaller-sized MAC Protocol Data Units (MPDUs). This fragmentation scheme is also adopted by the IEEE 802.15.3 MAC standard. Hereinafter, for simplicity, the fragmentation process is mentioned based on the WLANs. For many applications, the size of the MSDU is often so large that fragmentation is indeed necessary. Holland et al. [21] proposed a frame prediction scheme. This scheme predicts the optimal frame size for the next transmission according to the BER under the expected channel quality. However, Holland et al. did not consider fragmentation and rate adaptation. Other studies [24-26] developed a scheme to choose the optimal fragment size based on channel information such as the achievable data rate and goodput. Therein, although each MSDU can be fragmented according to the channel information, the size of the MPDUs remains unchanged during the MSDU transmissions and the transmission scheme is still static in nature, although a certain degree of optimization is performed. Moreover, in these schemes [24-26], the mechanism for exchanging the channel information is not clearly elaborated. Other fragmentation schemes without rate adaptation were proposed [27-29]. The fragmentation threshold is halved for each transmission failure during the transmission bursts [27], while it is doubled for each successful transmission and halved for each transmission failure [28]. The fragmentation size is tuned [29] to allow a fragment to fit in a dwell time in the frequency hopping communication system.

We propose a new receiver-based MAC protocol based on dynamic fragmentation. The proposed protocol is similar to RBAR and OAR. However, instead of allowing the transmission of multiple packets with a high data rate, a larger MPDU size is allowed, to reduce the overhead caused by the transmission of multiple fragments when the channel

condition is good. In addition, the proposed scheme adapts the fragment size during the MSDU transmission period, based on channel condition information obtained from the preceding fragment transmission. A fragment is generated on the fly, from the remaining MSDU, only when a fragment is ready for transmission, in contrast with the one-time fragmentation for MSDU used in other protocols (such as IEEE 802.11 MAC). Since the fragmentation process in the IEEE 802.15.3 standard is the same as that in the IEEE 802.11 standard as mentioned above, the proposed scheme is applicable to the MAC protocol in IEEE 802.15.3.

2.2 Preliminary Research

2.2.1 Distributed Coordination Function (DCF) in IEEE 802.11 MAC

The DCF mode in IEEE 802.11 MAC is called Carrier Sense Multiple Access/ Collision Avoidance (CSMA/CA), which is widely used in wireless LANs. In CSMA/CA, a node having a frame to transmit senses the channel for a DCF InterFrame Space (DIFS) idle time to check whether the channel is idle. If the channel is busy, the node defers the transmission until the channel is idle. When the channel is idle during a DIFS idle time, the node chooses its random backoff time and keeps sensing the channel during the chosen time period. The backoff timer decrements the chosen time as time goes on. If the channel remains idle when the backoff timer reaches zero, the node sends an RTS frame and the intended receiving node sends a CTS frame back to the sender after a Short InterFrame Space (SIFS) idle time. Because the RTS and CTS frames contain information about the duration of the incoming data transmission, other nodes overhearing the RTS or the CTS frame defer their transmissions for the duration defined by the Network Allocation Vector (NAV). This is known as Virtual Carrier Sensing, which prevents collision during data transmission. After receiving the CTS, the sender

transmits a data fragment and the receiver sends an ACK back to the sender after an SIFS idle time. The timing of the protocol used in DCF consists of cycles starting from the DIFS idle period and ending with the ACK.

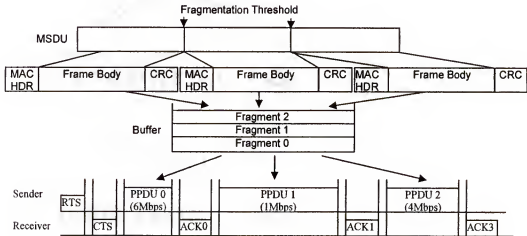


Figure 2-1. Conventional fragmentation process and the timeline of data transmission with rate adaptation

2.2.2 Fragmentation in IEEE 802.11

Fragmentation is the process of dividing a long frame into short frames. Figure 2-1 shows the fragmentation process in IEEE 802.11 MAC [4]. When an MSDU is passed down from the LLC layer, if the size of the MSDU is greater than the fragmentation threshold ($\text{aFragmentationThreshold}$), it is divided into smaller fragments. Each fragment, namely an MPDU, becomes a MAC layer frame with a MAC header. Then, a Physical Layer Convergence Protocol (PLCP) header and a preamble are added to the MPDU. The resulting frame is called a PLCP Protocol Data Unit (PPDU), which is a frame the transmitted over the air by the physical-layer. Fragmentation can be used to improve the transmission reliability in hostile wireless environments, because the probability of successful transmission increases as the size of MPDU decreases. Usually,

in IEEE 802.11 wireless LANs, an MSDU is fragmented into equal-sized MPDUs except for the last MPDU before the transmission attempt. These MPDUs are put into the buffer at the transceiver, and none of them will be refragmented further. All fragments are sent independently, and each is acknowledged separately. Once a sender contends for and seizes the medium, it will continue to send fragments with SIFS-sized gaps between the ACK reception and the start of the next fragment transmission, until either all the fragments of the MSDU have been sent, or an ACK is not received. When the transmission of a fragment fails, the contention process begins after a Distributed InterFrame Space (DIFS) idle time period. The remaining fragments are transmitted when the node seizes the channel again through the contention process. The transmission process for the fragments of an MSDU is called a fragment burst. Since the header of each MAC frame contains the information that defines the duration of the next transmission, the nodes that overhear the header update the NAV value for the next fragment transmission.

2.2.3 Rate-Adaptive Protocol Specified in IEEE 802. 11 MAC DCF Mode

Certain MAC protocols proposed [21-23] use RTS/CTS frames to exchange the selected data rate during the data transmission period. The receiver uses RTS to carry out channel estimation and rate selection. The selected rate is then fed back to the sender via CTS. The RTS and CTS packets are exchanged at the base rate to make sure that all nodes in the radio range can receive them without error. The performance evaluation of these protocols only considers the case when the MSDU size is less than `aFragmentationThreshold` (i.e., each node has only one fragment to send in its respective fragment burst). Since all MPDUs are of the same size when using the fragmentation scheme described above, the size of a data PPDU varies according to the selected rate.

Therefore, the duration of data transmission varies, as shown in Figure 2-1. Therefore, because of the variable duration of the data transmission, the duration value in the RTS frame is not the same as the actual transmission duration of the data frame. This causes a NAV update problem for nodes overhearing the RTS and MPDUs. Thus a two-step process is proposed in the RBAR protocol for NAV update. When the nodes overhearing the CTS packet update the NAV value with the duration calculated from the selected rate, the other nodes overhearing the RTS packet update the NAV value with a tentative duration, which is the duration for transmitting the MPDU at the lowest rate. When the nodes overhearing the RTS packet hear the MPDU, the NAV value is updated to the duration calculated from the rate in the PLCP header of the MPDU.

2.3 Proposed Protocol

2.3.1 Fragmentation Scheme

A new dynamic fragmentation scheme is proposed to enhance throughput under time-varying wireless environments. The proposed scheme contains the following key changes comparing to IEEE802.11 MAC:

- The transmission durations of all fragments, except the last fragment, in the physical-layer are set to be the same regardless of the data rate.
- Different *aFragmentationThresholds* for different rates are used based on the channel condition (i.e., a Rate-based Fragmentation Thresholding (RFT) scheme is used).
- A new fragment is generated from the fragmentation process only when the rate is decided for the next fragment transmission, resulting in Dynamic Fragmentation (DF).

In IEEE 802.11, with a single *aFragmentationThreshold*, the sizes of fragments are equal regardless of the channel condition. Therefore, the channel access time for a fragment varies with respect to the selected rate. For example, the channel access time for

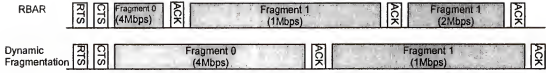


Figure 2-2. Timelines for RBAR and the proposed dynamic fragmentation scheme

a fragment at the base rate is longer than that for a fragment at a higher rate. It is generally assumed that the channel remains unchanged during the transmission of a fragment at the base rate. Thus, more data frames can in fact be transmitted when a higher rate is used in the same duration provided that the SNR is high enough to support the higher rate. From this observation, the OAR protocol [23] allows a node to a multi-packet transmission once it accesses a channel. However, multi-packet transmission has a higher overhead because of the additional MAC headers, PHY headers, preambles in data and ACK, and SIFS idle times. To overcome the shortcoming of multi-packet transmission, the proposed scheme fixes the time duration of all data transmission except for the last fragment.

To better understand the mechanism, Figure 2-2 shows the protocol timelines for the RBAR scheme and our dynamic fragmentation scheme. To generate fragments with the same time duration in a physical-layer, the number of bits in a fragment should be varied based on the selected rate. Thus, it is necessary to have different aFragmentationThresholds for different data rates. When the sender receives the selected rate from the receiver, the next fragment is then generated from the fragmentation process according to the aFragmentationThreshold for that rate. Thus, the fragmentation threshold at rate R is

$$Threshold_R = Threshold_B \cdot \frac{R}{B}, \quad (2-1)$$

where $Threshold_B$ is the aFragmentationThreshold at the base rate B and its unit is bit. At rate R , to transmit the same amount of information contained in a MPDU in the dynamic fragmentation scheme, the additional overhead in the single aFragmentationThreshold ($Threshold_B$) scheme is

$$Overhead = (T_{pre} + T_{PHY\_hdr} + T_{MAC\_hdr} + 2 \cdot T_{SIFS} + T_{ACK}) \cdot \left(\left\lceil \frac{R}{B} \right\rceil - 1 \right) \quad (2-2)$$

where T_{pre} , T_{PHY_hdr} , T_{MAC_hdr} , and T_{ACK} are time durations of the preamble, PHY header,

MAC header and ACK frame, and T_{SIFS} is the SIFS idle time. In Equation 2-2, $\left\lceil \frac{R}{B} \right\rceil$

indicates the number of data MPDUs that are needed in the single

aFragmentationThreshold scheme to transmit the same amount of data that one MPDU in our fragmentation scheme has. From Equation 2-2, we observed that higher data rate requires larger overhead.

In the fragmentation process in IEEE 802.11 MAC, a MSDU is fragmented into equal-sized fragments, which remain unchanged until all fragments in the burst are transmitted. If the channel quality is constant during the transmission of the fragment burst, the target PER can be met. However, this is not guaranteed in a wireless LAN because of two reasons. The first reason is that different fragments of the burst experience different level of channel quality because of the time-varying nature of the wireless channel. The second reason is that after the transmission of a fragment fails, the sender contends for the channel again to transmit remaining fragments, thus the channel quality is not guaranteed to be the same as that at the time when the previous fragment is transmitted. To achieve the target PER, both the data rate and a fragment size should vary according to the changing channel condition. Moreover, to better match the varying

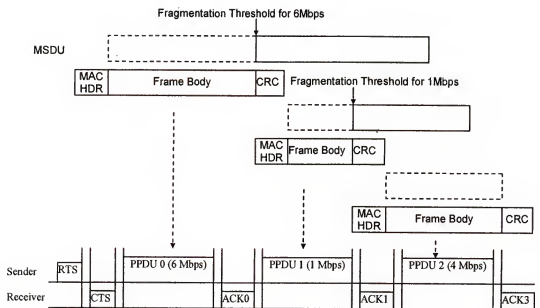


Figure 2-3. Dynamic fragmentation process and the timeline of data transmission

channel condition, instead of generating all fragments before transmitting the first fragment, each fragment should be generated at the time when the rate is chosen for the next transmission. As a result, the fragments in a burst may not be of the same size. Figure 2-3 illustrates the process of the proposed dynamic fragmentation scheme. When the transmission of a fragment fails, the size of the retransmitted fragment may not be the same as that of the originally transmitted fragment since the channel condition may have changed. When the fragment number of the most recently received fragment is the same as that of the already received fragment, the receiver discards the old fragment. Hence, the MSDU size is reduced only when a fragment is transmitted successfully (i.e., the sender receives an ACK from the receiver).

2.3.2 Rate-Adaptive MAC Protocol for Fragment Burst

With fragment burst transmission and rate adaptation for each fragment, data and ACK frames also participate in the rate adaptation process in the same way as RTS/CTS frames do. To support the rate adaptation process of a fragment burst, the physical-layer

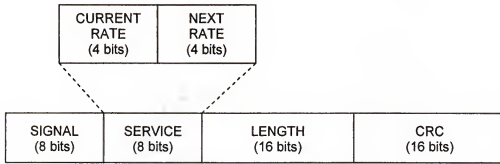


Figure 2-4. Physical-layer header format in the proposed protocol

header is modified as shown in Figure 2-4. The SERVICE field in the PLCP header is divided into two 4-bit subfields, namely the current rate and next rate subfields. The *current rate* subfield indicates the data rate of the current frame, whilst the *next rate* subfield indicates the selected data rate for the next incoming data frame. The values of two subfields in PLCP headers for RTS and data frames are the same because the *next rate* subfields in these headers indicate rates of frames transmitted from the receiver. After a sender sends a RTS frame at the base rate, a receiver estimates the channel and sends back a CTS frame to the sender with the selected rate stored in the *next rate* subfield. The sender modulates the fragment with the rate and sends a data frame to the receiver. After receiving the frame, the receiver predicts the channel condition for the next data frame and sends an ACK frame to the sender with the selected rate.

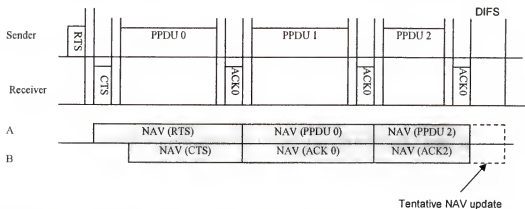


Figure 2-5. NAV update process in the proposed protocol

2.3.3 Network Allocation Vector (NAV) Update

In the proposed dynamic fragmentation scheme, the NAV update process is simpler than that in the RBAR protocol as described in Section 2.2. Figure 2-5 explains the NAV update process in the proposed protocol. Since the durations of all fragments in a fragment burst, except for the last fragment, are the same regardless of the data rate, an overhearing node can update the NAV value to the predefined duration when the More Fragments bit in the MAC header is set to 1. For the last fragment whose More Fragments bit is set to 0, the two-step process for NAV update proposed in the RBAR applies. At first, an overhearing node updates the NAV value with the duration of the normal data frame. This is called a tentative update as shown in Figure 2-5. When the last fragment and ACK frame are received, the NAV value is changed to the duration value in the MAC header since the duration values of the MAC headers in the last fragment and the ACK frame indicate the duration of the current transmission.

2.4 Simulation Setting

2.4.1 Wireless Channel Model

To reflect the fact that the surrounding environmental clutter may be significantly different for each pair of communication stations with the same distance separation, we use the log-normal shadowing channel model [57]. The path loss $\overline{PL}$ at distance d is

$$\overline{PL}(d)[dB] = \overline{PL}(d_0)[dB] + 10n \log\left(\frac{d}{d_0}\right) + X_\sigma, \quad (2-3)$$

where d_0 is the close-in reference distance, n is the path loss exponent and X_σ is a zero-mean Gaussian distributed random variable (in dB) with standard deviation σ (in dB).

We set n to 2.56 and σ to 7.67 according to the result of measurements for a wideband microcell model [57]. To estimate $\overline{PL}(d_0)$, we use the Friis free space equation

$$P_r(d_0) = \frac{P_t G_t G_r \lambda^2}{(4\pi)^2 d_0^2 L}, \quad (2-4)$$

where P_t and P_r are the transmit and receive power, G_t and G_r are the antenna gains of the transmitter and receiver, λ is the carrier wavelength, and L is the system loss factor which is set to 1 in our simulation. Most of the simulation parameters are drawn from the data sheet of Cisco 350 client adapter [58] (e.g. the output power, antenna gain, and so on). Finally, the long-term signal-to-noise ratio is

$$SNR_L[dB] = P_t - \overline{PL}(d) - N + PG[dB], \quad (2-5)$$

where N is the noise power which is set to -95 dBm [20]. In Equation 2-5, PG is the spread spectrum processing gain given by

$$PG[dB] = 10 \cdot \log_{10} \left(\frac{C}{B} \right), \quad (2-6)$$

where C is the number of chips per a symbol and B is the number of bits per a symbol.

We assume the signal format in IEEE 802.11b are employed. The numbers of chips per a symbol are 11 chips for 1 and 2 Mbps and 8 chips for 5.5 and 11 Mbps. The PG s for 1, 2, 5.5, and 11 Mbps are 10.4, 7.4, 3, and 0 dB [59], respectively. For 11 Mbps, since 8 information bits are encoded into 8-chip sequence, there is no spreading gain. We evaluate the performance of the proposed scheme in a time-correlated fading channel. The received SNR_L is varied by the Ricean fading gain α , which is generated according to the modified Clack and Gans fading model [60]. The Ricean fading gain α is complex Gaussian with mean $\sqrt{K/(K+1)}$ and variance $1/(2(K+1))$, where $K \geq 0$ is the Ricean

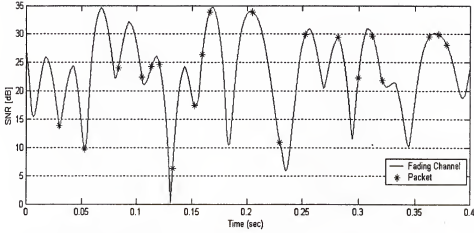


Figure 2-6. Packet arrival time on the fading channel

parameter, defined as the ratio of direct to diffuse power in the received signal. Under this model, the SNR of the received signal is

$$SNR[dB] = 20 \cdot \log_{10} \alpha + SNR_L[dB]. \quad (2-7)$$

The time varying nature of the wireless channel is described by the Doppler spread and coherence time, which are inversely proportional to one another. In our simulations, we consider the effect of the node speed on the change of the Doppler spread and coherence time. All nodes are assumed to move with the same speed. The maximum node speed in our simulation is 7m/s. Figure 2-6 shows the instances of data frame transmissions of one node over a time-correlated Ricean fading channel. In our simulator, pre-computed data set for fading gain is used as suggested by Punnoose et al. [60]. Each data in the set is applied to Equation 2-7 for constant time duration. Once a node accesses the channel, the starting point on the data set is randomly chosen. Whenever the constant time duration is passed, the next data from the set is applied to the channel. For the channel condition estimation and prediction in our simulations, we use the SNR measured at the end of the reception of RTS and data frames for the next fragment

transmission. In practice, more practical prediction and tracking algorithms are needed (e.g., the adaptive long-range prediction scheme [61-63]). However, the performance of the proposed scheme over existing prediction errors is also evaluated.

2.4.2 Network Environment

We assume that all nodes are uniformly distributed in space and within the radio range of each other so that the hidden and exposed terminal problems are not considered. The maximum distance between any two nodes is limited to 300 m, which is the maximum effective transmission range as indicated by Cisco system [58]. For simplicity, we assume that the PHY and MAC headers of all types of frames are modulated at the base rate and always reliably received. Since the control frames such as RTS, CTS, and ACK frames are much shorter than data frames, no transmission failure of these frames are considered in the simulation. The parameters used in this simulation studies are shown in Table 2-1. The choice of these parameters is based on the IEEE 802.11b DSSS standards.

Table 2-1. Simulation parameters based on IEEE 802.11b DCF mode

Parameter	Value
CWmin	31
CWmax	1023
SIFS time	10 us
DIFS time	50 us
Slot time	20 us
MAC header	272 bits
PHY header	48 bits
Preamble	144 us
ACK frame length	112 bits
RTS frame length	160 bits
CTS frame length	112 bits

To demonstrate the ability of the proposed protocol to adapt to the changing channel condition, we assume that the system adapts the data rate by properly choosing one from a set of modulation schemes according to the channel condition. The set of modulation schemes used in this simulation are DBPSK, DQPSK, 5.5 Complementary Code Keying (CCK), and 11CCK as defined in the standard [64]. One of the modulation schemes is chosen so that a target PER (packet error rate) can be achieved at the current channel SNR level. For simplicity, we will refer the PPDU as a packet in Section 2.4 as being commonly used in a physical-layer research community. The base data rate is set to 1 Mbps and the *aFragmentationThreshold* at the base rate is set to 800 octets cf. [65]. Thus, the number of symbols, N , in a MPDU, except for the last one, is set to 6400. However, the symbol rates according to the modulation type are different. The symbol rate for 1 and 2 Mbps is 1 Million Symbols per second (MSps) and that for 5.5 and 11 Mbps is 1.375MSps as shown by Fainberg [59] and Pearson [66]. As a consequence, the N s are 6400 for 1 and 2 Mbps, and 8800 for 5.5 and 11 Mbps. The SER equations for determining the SNR are found [67] over an additive white Gaussian channel. Since our simulations are performed over the slow fading channel scenario, the channel slowly changes within a packet. Therefore, we can assume that the symbol errors within a packet are approximately independent and use the SERs over the additive white Gaussian channel. For DBPSK,

$$SER = \frac{1}{2} e^{-E_s / N_0}, \quad (2-8)$$

where E_s / N_0 is the SNR per symbol. The approximated SER for DQPSK found [68] is given by

$$SER \leq 2 \cdot Q\left(\sqrt{\frac{E_s}{N_0}}\right). \quad (2-9)$$

The CCK is a variation of M-ary BiOrthogonal Keying (MBOK) modulation [59, 69]. The SER [67] is

$$SER = 1 - \frac{1}{\sqrt{2\pi}} \cdot \int_X \left(\frac{1}{\sqrt{2\pi}} \cdot \int_{(v+X)}^{v+X} e^{-x^2/2} dx \right)^{M/2-1} \cdot e^{-v^2/2} dv, \quad (2-10)$$

where $X = \sqrt{\frac{2E_s}{N_0}}$, and M is 4 for 5.5 Mbps, and 8 for 11 Mbps.

Based on the independent symbol error approximation, the PER is related to the SER by

$$PER = 1 - (1 - SER)^N, \quad (2-11)$$

where N is the number of symbols in a data packet. We set the target PER to 8% according to the IEEE 802.11 standard [4].

By consulting the SER performance curves calculated from Equation 2-8 to Equation 2-10 in Figure 2-7, the SNR ranges for the corresponding modulation schemes that the target SER is satisfied are, respectively,

$$R = \begin{cases} 1 \text{ (DBPSK)} & , \quad SNR < SNR_2 \\ 2 \text{ (DQPSK)} & , \quad SNR_2 \leq SNR < SNR_{5.5} \\ 5.5 \text{ (CCK)} & , \quad SNR_{5.5} \leq SNR < SNR_{11} \\ 11 \text{ (CCK)} & , \quad SNR_{11} \leq SNR \end{cases} \quad (2-12)$$

where SNR_i is the SNR threshold for the data rate i to meet the target SER. The SNR ranges for the conventional fragmentation scheme described in Section 2.2 are different from those of the proposed dynamic fragmentation scheme. While the target SERs are the same for all data rates in our fragmentation scheme, the target SERs at different data

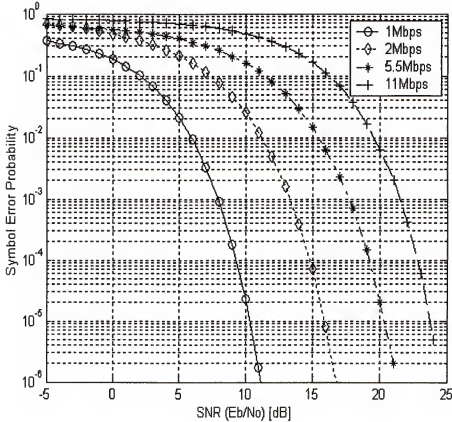


Figure 2-7. Symbol error rates of DBPSK, DQPSK, 5.5CCK, and 11CCK

rates for the conventional fragmentation there are different because the number of symbols in a data packet changes for different data rates so that different SNR ranges are needed to meet the same target FER requirement. Thus two sets of SNR ranges are used for the two fragmentation schemes.

According to the IEEE 802.11 standard, the maximum MSDU size is 2304 octets. However, because we consider the case of bulky data traffic where an IP packet can be as large as 64 koctets, we simulate with a larger maximum MSDU size than the maximum MSDU size in IEEE 802.11. Thus we assume that the MSDU size is uniformly distributed over the range from 2304 octets to 6000 octets at each node. In addition, we assume that the MSDUs at any node are always available. In the IEEE 802.11 standard,

the value of dot11MAXTransmitMSDULifetime is 512ms for the 2304-maximum MSDU size. Because our simulation uses MSDU sizes larger than 2304 octets, dot11MAXTransmitMSDULifetime increase in proportion to the rate of 6000 octets and 2304 octets. Thus dot11MAXTransmitMSDULifetime is set to 1.3s. The Station Long Retry Time (SLRC) is set to 7. All simulations are performed for 300 seconds simulation time.

We compare the performance achieved by three different configurations:

- **Case 1:** Rate-based Fragmentation Thresholding with the proposed Dynamic Fragmentation scheme (RFT-DF);
- **Case 2:** Rate-based Fragmentation Thresholding with the Conventional Fragmentation scheme (RFT-CF); and
- **Case 3:** Single Fragmentation Threshold with the Conventional Fragmentation scheme (SFT-CF).

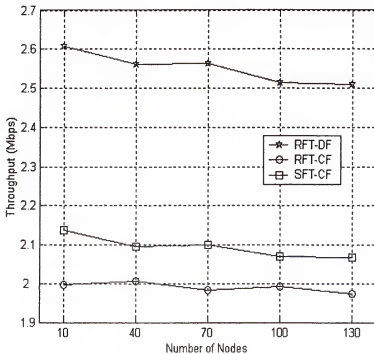


Figure 2-8. Throughput as a function of number of nodes

2.5 Performance Evaluation

2.5.1 Impact of the Number of Nodes

Figure 2-8 shows the throughputs obtained by these three configurations with Ricean parameter $K = 2$ and 4m/s node speed as the number of nodes increases from 10 to 130 with step size 30. From Figure 2-8, we observe that the throughput of RTF-DF is up to 22% higher than that of SFT-CF and 30.6% higher than that of RTF-CF. Moreover, we observe that increasing the number of nodes from 10 to 130 causes 3.7% degradation in throughput. The idle time caused by one collision is the sum of the backoff time, RTS/CTS transmission time, one SIFS, and one DIFS. This idle time duration is small compared to the lost time caused by data packet errors. In addition, in the fragment burst transmission, the channel access time of a node is longer than that of a single data packet transmission because once the node gains channel access, it transmits several fragments without any further contention. Thus, the effect of collisions due to the larger number of nodes on the throughput is small. In the simulation, we observe that only 6.7% of the total simulation time in RFT-DF with 130 nodes is caused by contention.

A detailed evaluation of the performance differences among three configurations is presented in Figure 2-9. Figure 2-9(a) shows the average numbers of packets per MSDU for the three configurations. The number of packets in SFT-CF is about three times of that in RFT-DF.

The time overheads relative to RFT-DF are shown in Figure 2-9(b). The time overhead relative to RFT-DF is defined as

$$RTO [\%] = \frac{TO - TO_{RFT-DF}}{TO_{RFT-DF}} \cdot 100, \quad (2-13)$$

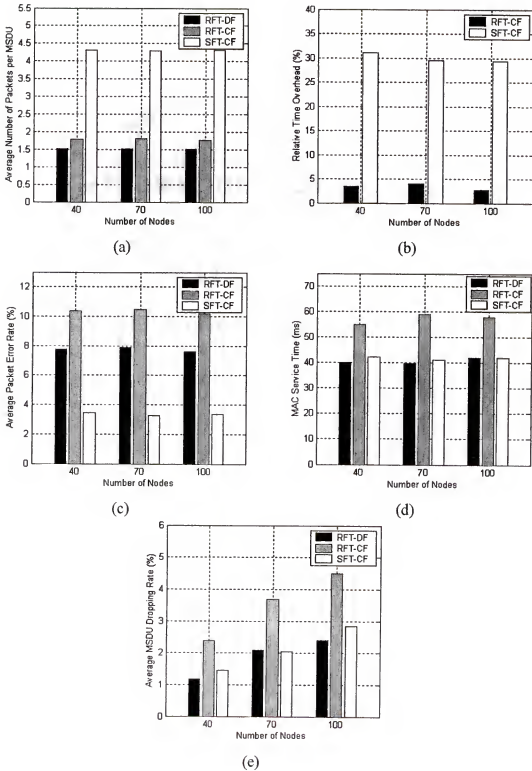


Figure 2-9. Performance evaluations for three schemes. A) Average number of packets per MSDU. B) Relative time overhead. C) Average packet error rate. D) MAC service time. E) Average MSDU dropping rate.

where TO and TO_{RFT-DF} are time overheads of the other configuration and RFT-DF, respectively. The time overheads are caused by backoff time, RTS/CTS/ACK frame transmissions, DIFS/SIFS idle times, preamble, and MAC and PHY headers. The time overhead of SFT-CF is much higher than, around 32%, that of RFT-DF. However, for the packet error rate, RFT-DF is higher than that of SFT-CF as shown in Figure 2-9(c). Although the SNR threshold is chosen to meet the target packet error rate, RFT-DF has higher FER than SFT-CF has because the large packet size in RFT-DF has a higher probability of experiencing channel change within the packet transmission.

Figure 2-9(d) shows the average MAC service time. We define MAC service time as the time duration for successful transmission of one MSDU (i.e., the time from the MSDU is ready for transmission to the MSDU is acknowledged for a successful transmission). Contrary to time overhead, the MAC service time includes transmission times of packets, elapsed times caused by packet errors, and waiting times caused by transmissions from the other nodes. We notice that although waiting times account for around 80% of the average MAC service time, they do not affect to the calculation of the throughput. The MAC service time of SFT-CF is about 5% higher than that of RFT-DF. Comparing to the result for time overhead, the difference between RFT-DF and SFT-CF is less significant in terms of MAC service time. This is because that the elapsed time caused by packet errors in RFT-DF is larger than that of SFT-CF. In addition, because the waiting times account for a large portion of the average MAC service time, the effect of the time overhead difference on the MAC service time reduces.

Finally, the average MSDU dropping rates are shown in Figure 2-9(e). The factors affecting MSDU dropping are `dot11MAXTransmitMSDULifetime` and `SLRC`. However,

we observe that $\text{dot11MAXTransmitMSDULifetime}$ is a main factor affecting MSDU dropping. The difference between RFT-DF and SFT-CF in terms of the average MSDU dropping rates is less significant similar to the difference between the MAC service times. We observe that a dropped MSDU has more packet errors than a successfully transmitted MSDU has. In addition, a transmission failure leads to additional waiting and idle times to access the channel. These times become longer as the number of nodes increases. Thus the MSDU dropping rates for the three configurations increase significantly with increasing the number of nodes.

2.5.2 Impact of the Ricean Parameters

Figure 2-10 shows the performance of the three configurations described above under different fading environments with 40 nodes and 4m/s node speed. The Ricean parameter, K , indicates the strength of the line of the sight component of the received

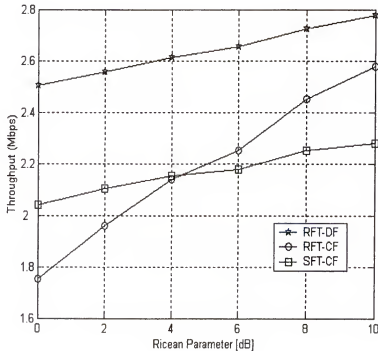


Figure 2-10. Throughput as a function of Ricean parameter, K

signal. For $K=0$, the channel has no line-of-sight (LOS) component, corresponding to the worst-case scenario, which is referred to as Rayleigh fading. As K increases, the strength of the LOS component increases. Therefore, the performances of the three configurations improve with increasing K values. From Figure 2-10, we observe that the throughput of RFT-DF is up to 21.8 % higher than that of SFT-CF at $K=10$ and 42.8% higher than that of RFT-CF at $K=0$. For $K > 4$, the performance gain of RFT-CF is up to 13.6% higher than that of SFT-CF. As the error rate of the pre-fragmented MPDUs reduces in the channel with the higher value of K , the gain due to the overhead difference overcomes the loss due to the packet errors.

2.5.3 Impact of Node Speed

We vary the speed of nodes, but assume that no node is out of the radio range. As

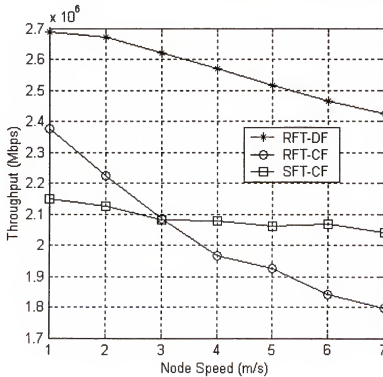


Figure 2-11. Throughput as a function of node speed

the speed of nodes increases, the coherence time of the channel reduces. This implies that the channel changes faster. Figure 2-11 shows the performance of the three configurations for 7 different speeds ranging from 1m/s (pedestrian speed) to 7m/s. The number of nodes is 40 and the Ricean parameter is $K = 2$. From Figure 2-11, we observe that the throughput of RFT-DF is 25.2% higher than that of SFT-CF at 1m/s node speed. The performances of RFT-DF and RFT-CF degrade faster than that of SFT-CF as the node speed increases. This can be explained as follows. As the node speed increases, the channel coherence time is shorter, hence the probability that the channel condition changes in the middle of a packet transmission is higher in the fragmentation scheme with rate-based fragmentation thresholding than in that with a single fragmentation threshold. However, the performance of RFT-DF is still 18.7% higher than that of SFT-

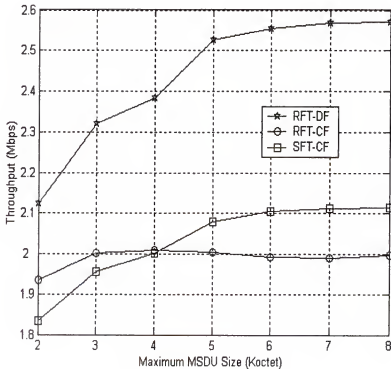


Figure 2-12. Throughput as a function of maximum MSDU size

CF at 7m/s node speed. In addition, we notice that when the node speed is less than 3m/s, the performance of RFT-CF is better than that of SFT-CF. When the nodes move at low speeds, the channel changes slowly enough, so that it remains constant during one MSDU transmission. At higher speeds, MPDUs fragmented previously in RFT-CF cannot cope with the channel change.

2.5.4 Impact of the Maximum MSDU Size

We also vary the maximum MSDU size from 3 koctets to 10 koctets with a step size of 1 koctets, and observe changes in the performance. According to the maximum MSDU size, *dot11MAXTransmitMSDULifetime* is set to the corresponding value. The number of nodes is 40, the Ricean parameter is $K = 2$, and the node speed is 4m/s. From Figure 2-12, we observe that the throughput of RFT-DF is around 21.6% higher than that

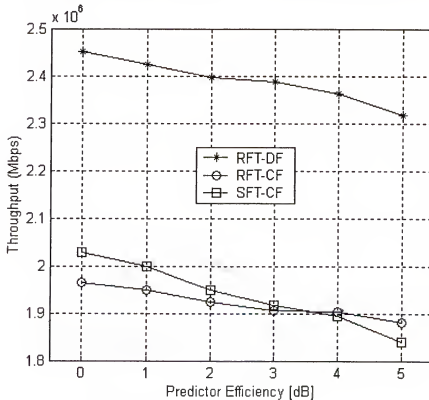


Figure 2-13. Throughput as a function of predictor efficiency

of SFT-CF. The performance gains of RFT-DF and SFT-CF increase with increasing maximum MSDU size. However, the performance of RFT-CF decreases at large MSDU sizes because the number of pre-fragmented MPDUs increases with larger MSDU. This makes RFT-CF harder to cope with the channel change.

2.5.5 Impact of the Channel Estimation Error

Here we evaluate how performance is affected by inaccuracy in the channel prediction process. The predicted channel gain, $\hat{\alpha}$, can be written as

$$\hat{\alpha} = \alpha + e, \quad (2-14)$$

where α is a true channel gain and e is an additive prediction error, which is assumed to be a white and zero mean complex Gaussian random variable [62, 63]. From studies of the prediction algorithms [61-63, 70], the prediction error reduces as SNR increases. In this simulation, the channel prediction error variance is obtained by

$$\sigma_e^2[dB] = \gamma[dB] - SNR[dB], \quad (2-15)$$

where γ , referred to as *predictor efficiency*, indicates the performance of the channel predictor. In the simulation, we vary the prediction efficiency from 0 to 5 dB. The value of 5dB corresponds to the worst scenario. The number of nodes is 40, the Ricean parameter is $K = 2$, and the node speed is 4m/s. From Figure 2-13, we observe that the throughput of RTF-DF is up to 25.8% higher than that of SFT-CF and 24.7% higher than that of RTF-CF. Here even with inaccurate channel prediction, the proposed scheme still achieves higher performance gain than the other schemes do.

2.6 Conclusion

We proposed a new rate-adaptive MAC protocol with dynamic fragmentation. The major innovation is the use of multiple fragmentation thresholds for different rates to

generate a new fragment from a (remaining) MSDU only after the rate for the next transmission is selected. With this scheme, the nodes with good channels can transmit more data than the ones with bad channels. In addition, the use of constant fragment transmission duration in the physical-layer simplifies the process of NAV update in our rate-adaptive system. Our results show that the proposed dynamic fragmentation scheme achieves throughput gain from 14.4% to 29% over the conventional fragmentation scheme used in the IEEE 802.11 MAC protocol.

CHAPTER 3 TWO-STEP MULTIPOLLING MAC PROTOCOL FOR CENTRALIZED WIRELESS LANS

3.1. Introduction

As described in Chapter 1, the MAC protocol in IEEE 802.11 [4] consists of two coordination functions: Distributed Coordination Function (DCF), described in Chapter 2, and Point Coordination Function (PCF). In the DCF as described in Chapter 2, a set of wireless stations (STAs) communicate directly with each other using a contention-based channel access method, namely, Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA). In the PCF, the channel access of each station is controlled by polling from a Point Coordinator (PC) at the access point (AC). While the DCF is designed for asynchronous data transmission, the PCF is mainly intended to transmit time-bounded services such as voice and video. The DCF and PCF can coexist by alternating the Contention Free Period (CFP), ruled by PCF, and the Contention Period (CP), ruled by DCF.

As the capacity of WLAN increases, it is also important to improve the quality of service (QoS) for real-time multimedia applications. Since the controlled channel access can reduce the time wasted in accessing the channel during the backoff process in the DCF, the PCF is an appropriate scheme for applications with QoS requirements in WLANs. However, in IEEE 802.11 MAC, the scheduling algorithm for a polling sequence is based on the Round-Robin scheme, which is not suitable for handling real-time applications with various QoS requirements. Furthermore, the polling scheme in the

PCF introduces significant overhead. The overhead increases the transmission delays on time-bounded traffics and wastes the scarce wireless channel bandwidth. The overhead is caused not only by the polling frames themselves (since one polling frame polls only one station at a time), but also by polling the stations with no frame to transmit.

Consequently, most studies of the PCF in WLAN [56-64] have focused on these two factors: the scheduling scheme and the overheads. The scheduling schemes [56-58] are proposed to support multimedia services. In these schemes, all traffic types are differentiated by priorities and the polling sequence is scheduled according to the priorities of the traffics. To reduce the overhead caused by the polling frames, multi-polling schemes are proposed [59-61]. The idea is to poll all stations in one shot by means of one polling frame, instead of polling one station at a time. Consequently, this can reduce the overheads due to the polling frames. The protocols [62, 63] aim to reduce the unnecessary polling frames used for stations with no pending frames to transmit based on statistical estimation of the traffic characteristic or information reported by a station during the CP. Further performance improvement is achieved by simply removing an acknowledgment (ACK) frame in the PCF [64]. Recently, the IEEE 802.11e Task Group (TG) has proposed an enhanced function, namely the Hybrid Coordination Function (HCF) [5], to support the QoS services. In the HCF, the PC is allowed to start the CFP at any time during the CP and the channel access in the CFP is controlled by the polling method in IEEE 802.11.

However, most proposed polling algorithms only consider a constant physical transmission rate. Since a typical wireless channel is time-varying and most wireless networks support several different data rates in the physical-layer, an efficient

communication system can be designed by selecting the data rate according to the channel condition as proposed [24-28, 32-38]. Nevertheless, the rate-adaptive polling-based MAC protocol for WLANs has not yet been investigated.

In this chapter, we propose an efficient, polling-based MAC protocol, Two-Step Multi-Polling (TS-MP), to support real-time applications. In this protocol, we use two multi-polling frames with different purposes. The first multi-polling frame is sent to collect information such as the number of pending frames at each station and the physical transmission rate of each communication link. Based on such information, the PC schedules a polling sequence for data transmission, which is then broadcast in the second multi-polling frame. The proposed protocol not only overcomes the deficiencies of existing polling schemes, but also helps to implement rate adaptation.

3.2. Preliminary Research

3.2.1 Point Coordination Function (PCF) of IEEE 802.11 MAC

IEEE 802.11 MAC defines a superframe structure as shown in Figure 3-1. The superframe consists of two time periods: CFP and CP. During the CFP, medium access is controlled by the PCF. The CFP begins with a beacon frame containing parameters needed to control the superframe. The protocol used in the PCF in IEEE 802.11 MAC is based on a polling scheme controlled by a PC in such a way that contention-free transmission is guaranteed. The PC keeps the list of stations registered in its Basic Service Set (BSS), which is the set of stations controlled by the PC. Each station can transmit its frame only when it is polled by the PC. The transmissions of frames in the PCF are shown in Figure 3-1. The PC polls one station at a time. Hereafter, the polling scheme used in the PCF is called Contention Free Single Polling (CF-SP). When the PC itself has a pending data frame to the station to be polled, it transmits the data frame by

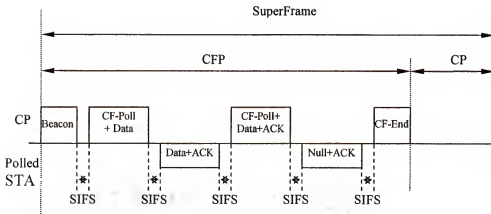


Figure 3-1. Channel Access of IEEE 802.11 PCF during CFP

piggybacking it into the polling frame. Moreover, if the PC needs to acknowledge a previously received frame, an ACK frame is also combined with the piggybacked polling frame. When the station receiving the frame from the PC has a pending frame, the data and ACK frames are similarly combined by piggybacking and transmitted back to the PC. When all stations in the polling list are polled or the CFP expires according to aCFPMaxDuration defined in the beacon frame, the PC sends a specific control frame, called Contention-Free (CF) End frame, to signal the end of the CFP. There is a Short InterFrame Space (SIFS) idle time between two consecutive frame transmissions in the CFP.

By reducing time consumption due to contention, the PCF is capable of supporting time-bounded services. However, for QoS provisioning, the following problems mentioned may arise:

1. The low throughput due to overhead induced by the polling frames
2. The inefficient Round-Robin scheduling algorithm
3. Lack of information such as the current number of pending frames in a station and the data rate in the physical-layer with respect to the channel conditions
4. Potential collisions caused by stations in a neighboring BSS

5. The unpredictable transmission time of a polled station.

3.2.2 Contention Free Period using Hybrid Coordination Function (HCF) in IEEE 802.11e MAC

The IEEE 802.11e TG proposes some enhancements to overcome the problems 4 and 5 (Section 3.2.1). The HCF proposed by the IEEE 802.11e TG controls transmissions of stations in the CFP as well as in the CP. The HCF in the CFP uses the CF-SP scheme in the PCF with two enhancements. The first one is the use of RTS/CTS handshaking between two communication stations as defined in the DCF of IEEE802.11 MAC. The exchange of RTS and CTS frames is performed after a station is polled and before data frame transmission is started. The stations overhearing the RTS or CTS frame in a neighboring BSS set their Network Allocation Vector (NAV) to the value in the RTS or CTS frame, and will not transmit during the time specified by the NAV. As a consequence, the polled station can transmit its data frame free of collision caused by stations in neighboring BSS.

The second enhancement is the use of Transmission Opportunity (TXOP). TXOP is the maximum time duration in which a polled station can transmit its frames. If at a polled station, the physical transmission rate is low and a pending frame size is long, the transmission time of the polled station will occupy a large portion of the CFP. For instance, according to the IEEE 802.11 standard, the maximum frame size is 2304 bytes and the lowest data rate in the physical-layer is 1 Mbps. In this case, the transmission time can be more than 20 msec. This long transmission time will reduce the number of stations that can be polled during the remaining time in the CFP. In the HCF, each station is assigned with a TXOP to prevent the transmission of any station from dominating the CFP.

3.2.3 Multipolling Schemes

Although the IEEE 802.11e TG enhances the polling scheme to mitigate some problems of the PCF, other problems, such as problem 1 to 3 (Section 3.2.1), still remain unaddressed. A number of MultiPolling (MP) schemes have been proposed to reduce the overhead due to the polling frames [59–61]. The first proposed multipolling scheme is Contention Free MultiPolling (CF-MP) [59]. In this scheme, the PC sends a multipolling frame with a polling sequence and time duration assigned to each station for frame transmissions after the beacon frame in the CFP. However, if a polled station does not have enough pending frames to utilize the assigned time duration, the remaining time is wasted. The polling scheme [60] focuses on the case when a polled station fails to receive a multipolling frame from the PC. To increase the reliability of receiving the polling information for all stations, each station sends its data frame appending the polling information. In this way, a station that fails to receive a multipolling frame from the PC has chances to obtain the polling information from the transmissions of other stations. Of course, this introduces additional overhead due to the redundant polling information.

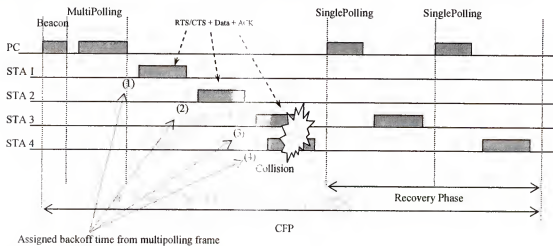


Figure 3-2. Sample scenario for CP-MP

Contention Period MultiPolling (CP-MP) proposed recently [61] applies the channel access scheme in the DCF of IEEE 802.11 to the PCF. After broadcasting the beacon frame, the PC sends a multipolling frame containing the transmission sequence, the allocated TXOPs and the initial backoff time for each station. Each backoff time must be unique. After receiving the polling frame, each station follows the rule of CSMA/CA. That is, each station reduces their backoff time, assigned by the PC, by one at a time if the channel is idle during a slot time. When the backoff time of a station reaches zero, the station sends its data frame. If it does not have a pending frame, it sends a Null frame. In order to avoid collision with the transmission from a station in a neighboring BSS, CP-MP uses RTS/CTS handshaking before a data frame transmission, as in the HCF. Because of the use of carrier sensing based channel access, there may be collisions even in the same BSS. It is assumed that all stations can hear or sense transmissions from the PC. However, it is not guaranteed that all stations in the BSS can hear or sense transmissions from all other stations. For instance, when a station cannot sense the transmission of the RTS or CTS frame, it transmits its data frame after its backoff time expires and this leads to collision. For the stations experiencing collision, the PC polls them using the CF-SP scheme after the last station in the polling sequence finishes its transmission. This time period is named recovery phase. Figure 3-2 shows a sample scenario of the operation of the CP-MP protocol in the CFP. In Figure 3-2, the polling sequence is Stations 1, 2, 3, and 4 and the backoff time assigned to each station is the same as the station number (e.g., 1 slot time for Station 1 and 4 for Station 4). Each station transmits its data frame when the backoff time is zero and the channel is idle. When Station 3 transmits its RTS frame, Station 4 also sends its RTS frame after one slot

time since Station 4 cannot hear the transmission from Station 3. Therefore, collision occurs. These collided stations are polled using single a polling frame in the recovery phase.

3.3. Two-Step Multipolling Scheme

3.3.1 Motivation

In Section 3.2, we present the problems of polling scheme in IEEE 802.11 MAC and some enhanced polling schemes to overcome these problems. Unfortunately, some of these schemes may actually aggravate some of the problems mentioned before. While the polling schemes in the HCF and the CP-MP scheme solve the collision problem caused by a station in a neighboring BSS, they introduce more overheads due to the RTS/CTS exchanges. In addition, while the CP-MP scheme reduces the overhead due to the polling frames, it may introduce collision between stations even in the same BSS. In addition, the scheduling scheme for the polling sequence and the TXOP allocation is not clearly specified in any of aforementioned polling schemes.

We now introduce a way for the PC to obtain information from each station in every CFP in order to efficiently schedule the polling sequence. In addition, we also consider a multipolling scheme to reduce overhead. The information may be the buffer status of each station and the physical transmission rate for each communication link in the current superframe. Utilizing this information, the PC can efficiently schedule the polling sequence and assign TXOPs to stations. In addition, it can reduce the overhead. We observe that in the current polling schemes, the PC must poll all stations regardless of whether a station has pending frames or not because the PC does not have any knowledge about the buffer status of each station. If the PC knows the buffer status of each station, a station without any pending frame should not be polled. In addition, when the physical

transmission rate for each communication link is known, the PC can estimate the channel access time for each station, which helps to determine the TXOP for the station. In particular, the rate adaptation scheme in the CFP can be designed. Many rate-adaptive MAC protocols for wireless networks have been proposed to adapt to time varying wireless channels. However, while the effect of the rate adaptation in the PCF is evaluated by Crow et al. [65], there are no rate-adaptive MAC protocols for the PCF proposed in the current literature.

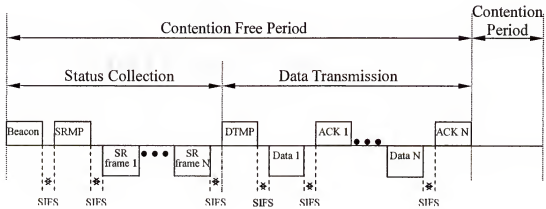


Figure 3-3. Time line of TS-MP Protocol during CFP

With these considerations, we propose a new polling-based MAC protocol in Section 3.3.2. Moreover, we present how the proposed protocol can overcome the problems discussed in Section 3.2.1.

3.3.2 TS-MP

We refer to the proposed multipolling scheme as Two-Step MultiPolling (TS-MP). We first implement an efficient scheduler and an appropriate TXOP allocation algorithm to obtain the required information from each station. We then introduce a rate adaptation mechanism to adjust the transmission rate according to the link information. And finally, we discuss how the overhead caused by polling frames can be reduced. The proposed

scheme overcomes the problems, such as a collision in a BSS, caused by other multipolling-based MAC protocols. Figure 3-3 illustrates the operation of the proposed TS-MP MAC protocol. The CFP period is divided into two sub-periods: Status Collection Period (SCP) and Data Transmission Period (DTP). A detailed operational description of these periods will be given in Section 3.3.2.1 and 3.3.2.2.

3.3.2.1 Status collection period

After broadcasting the beacon frame at the beginning of the CFP, the PC transmits the first multipolling frame, called Status-Request Multi-Poll (SRMP), to collect information from each station. Figure 3-4(a) shows the frame structure for the SRMP, whose length varies with the number of stations to be polled. The Polling Count subfield indicates the number of stations to be polled and the AID subfield is an association identifier, which identifies a station in the BSS. The stations to be polled are selected by the first scheduling scheme explained in Section 3.3.3. Each station polled by the SRMP

Byte : 2	6	1	2	...		2	4
Frame Control	BSSID	Polling Count (N)	AID 1			AID N	FCS

(a)

Byte : 2	6	2	1	2	4
Frame Control	BSSID	Tentative NAV	Buffer Status	AID	FCS

(b)

Byte : 2	6	1	5 x Polling Count (N)			4
Frame Control	BSSID	Polling Count (N)	Polling Control			FCS
			AID (2 bytes)	Rate (1 byte)	TXOP (2 bytes)	

(c)

Figure 3-4. Frame structures. A) SRMP. B) SR. C) DTMP.

frame sends a Status-Response (SR) frame back to the PC with some status information. Figure 3-4(b) shows the frame format for the SR frame. Specially added fields in the SR frame are the Tentative-NAV and Buffer Status fields. The Tentative-NAV field indicates the tentative time duration used for NAV allocation of stations belonging to a neighboring BSS that hear the transmission of the sender. In order to avoid the collision caused by the transmission of a station in the neighboring BSS, when a station hears a SR frame with different BSSID number, it sets its NAV to the value in the Tentative-NAV field and does not transmit during the period of the NAV. When the station in a neighboring BSS receives a data frame from the same station, the NAV value is reset to the value in the Duration field of the data frame. The value of Tentative-NAV field may indicate the end of the CFP. If a polled station does not have a frame to transmit in this CFP, the value of the Tentative-NAV field is set to zero since this station will not be polled for a data frame transmission at the second multipolling period. The Buffer Status field indicates the number of pending frames in the buffer of a station. This information is important for the PC to schedule a polling sequence and set the TXOP for each station in the incoming data transmission period. Moreover, the information about the pending frames reduces the time loss due to the polling of stations with no pending frames because these stations are removed from the polling sequence for the data transmission.

3.3.2.2 Data transmission period

After receiving the last SR frame, the PC sends a Data Transmission Multi-Poll (DTMP) frame. A polling sequence in the DTMP is constructed by the second scheduler based on the information obtained from the SR frames. The operation of this scheduler will be described in Section 3.3.3. The frame format is illustrated in Figure 3-4 (c). The Polling Count field is the number of stations to be polled in the Data Transmission Period

(DTP). The Polling Control field consists of three sub-fields: AID, TXOP, and Rate.

These three sub-fields specify the ID of a station to be polled, the time duration assigned to a station for transmission of pending frames, and the data rate for uplink frames, respectively. After the PC estimates the channel with the received SR frames in the SCP, a data rate is chosen to the transmission of the polled station. In comparison to the inaccurate TXOP allocations in the HCF and CP-MP, the PC can accurately allocate TXOPs to stations based on the information such as the number of polling frames and the physical transmission rate for these frames, which are obtained from SR frames.

Therefore, the time waste due to the inaccurate allocation of TXOP discussed in Section 3.2 is reduced. Each polled station transmits data frames with the given data rate from the DTMP frame after the predecessor's TXOP expires. There is a Short Interframe Space (SIFS) idle time between two consecutive TXOPs.

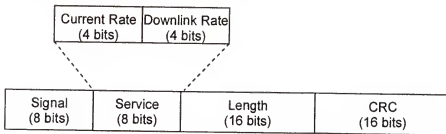


Figure 3-5. PLCP header format for TS-MP

3.3.2.3 Rate adaptation

To support rate adaptation in the CFP, the physical-layer header is modified as shown in Figure 3-5. The Service field in the physical-layer header is divided into two 4-bit subfields, namely the Current Rate and Downlink Rate subfields. The Current Rate subfield indicates the data rate of the current frame and the Downlink Rate subfield indicates the data rate selected through the channel estimation based on the received SRMP frame at a station. The value in Downlink Rate subfield is used to generate the

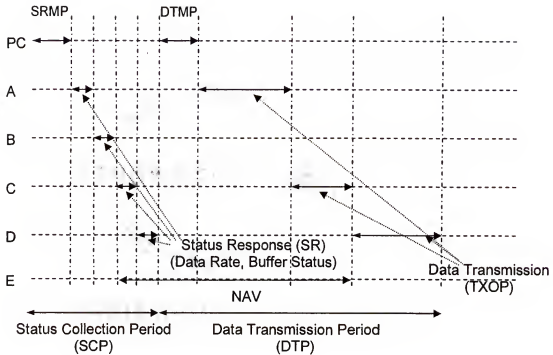


Figure 3-6. Example of TS-MP protocol

downlink frame at the PC. For instance, after the PC sends the SRMP frame at the base rate, each polled station estimates the channel and sends the SR frame containing the selected rate in the Downlink Rate subfields back to the PC. The SR frame is transmitted at the base rate. If the PC has a pending frame to transmit, the frame is modulated and coded according to the data rate informed by the SR frame of the destination station. For any frame from the PC, the value of Downlink Rate subfield is set to zero, which does not indicate any data rate because the subfield is used only by the SR frames. The data rates for the uplink data frames are informed by the DTMP frame after the communication links are estimated based on the SR frames at the PC. Using this operation, the physical transmission rate for the uplink and downlink transmissions can be dynamically adjusted according to the current channel condition.

3.3.2.4 Sample scenario

Figure 3-6 shows a timing diagram to illustrate the operation of the proposed multipolling MAC protocol. We assume that there are one PC and four real time traffic stations (A, B, C, and D) in the BSS. Station E is a station in a neighboring BSS and can hear the transmission from Station C. In the beginning of the CFP, The PC sends a SRMP frame with the transmission sequence $A \rightarrow B \rightarrow C \rightarrow D$. After the SRMP transmission and SIFS, each station sends a SR frame back to the PC. When Station E hears the SR frame from Station C, it sets its NAV to the value of the Tentative-NAV field. In this example, it is assumed that station B does not have any frame to transmit. As a consequence, the station B is removed in the polling sequence in the DTMP frame. All stations except Station B, start to transmit according to the sequence given in the DTMP frame, and their physical-layer frames are generated using the data rates specified in the DTMP. When Station E hears the transmission of the data frame from Station C, it resets its NAV to the value of the Duration field in the MAC header.

3.3.3 Polling Scheduler

As mentioned in Section 3.2, the commonly used scheduling method for the CFP in WLANs is the Round-Robin scheme, which is not efficient in dealing with services with various QoS requirements. To design a better scheduling algorithm, the PC needs to have information about the node status and the channel condition before polling all stations. Using the proposed protocol described in Section 3.3.2, the PC is able to obtain information needed for scheduling a polling sequence and as a consequence a better scheduling scheme can be implemented

Before describing two proposed scheduling schemes, we introduce two main factors that affect the scheduling process The first factor is the Service Period of Station i,

SP_i . The SP_i is the estimated inter-arrival time of frames at Station i with payload P_i which is given by

$$SP_i = \left\lceil \frac{P_i \cdot 8}{M_i \cdot T_{SF}} \right\rceil, \quad (3-1)$$

where P_i , M_i and T_{SF} are the payload in the MAC frame in bytes, the average data arrival rate in the MAC layer at node i , and the time duration of a superframe, respectively. P_i and M_i are obtained from the admission control unit in the PC during the association period. As shown in Equation 3-1, SP_i is expressed in the number of superframes and is also calculated by the admission control unit in the PC during the association period. We define another parameter ω_i related to SP_i in order to manage the polling time of Station i . The polling time will be illustrated in detail in Section 3.3.3.1. ω_i is initialized to be SP_i and decreased by one every superframe passed until it reaches to one. When ω_i becomes one, it is reset to SP_i at the next superframe.

The second factor is E_i^k , which is the normalized number of transmitted frames during the previous W superframes at Station i in superframe k . This parameter can be defined as

$$E_i^k = \sum_{j=k-W}^{k-1} e_i^j / M_i, \quad (3-2)$$

where W , the averaging window size, is the number of previous superframes to be considered for the averaging, and e_i^j is the number of transmitted frames in the j th superframe at Station i in the averaging window. This parameter is tracked and updated by the PC in every superframe.

3.3.3.1 First scheduler for SRMP

A scheduler for SRMP is useful for the case when there are many stations to be polled within the limited CFP period. When the number of stations to be polled by SRMP is very large, a large amount of time is spent during SCP, leading to excessive overhead and poor performance of the polling scheme. In order to avoid this situation, the following scheduler for SRMP is proposed:

- **Step 1:** Determine the number of stations to be polled in SRMP

The number of stations to be polled in the current CFP is determined from the information obtained in the previous CFPs. When the PC experiences a shortage of DTP to poll all stations with frames to transmit in the previous CFP, the number of stations to be polled in SRMP is reduced by one. The number is increased by one when DTP is enough to poll all stations with frames to transmit in the previous CFP. The number of stations to be polled for SRMP in the j th CFP can be expressed as follows:

$$N_j = \begin{cases} N_{j-1} - 1, & \text{if } T_{alloc}^{j-1} > T_{DTP}^{j-1} \\ N_{j-1} + 1, & \text{otherwise} \end{cases}, \quad (3-3)$$

where T_{DTP}^{j-1} is the time duration of DTP in the previous CFP and T_{alloc}^{j-1} is the sum of TXOPs estimated for all stations in the previous CFP as follows:

$$T_{alloc}^{j-1} = \sum_{i=1}^n T_{TXOP}^i. \quad (3-4)$$

- **Step 2:** Design the polling sequence

Once the number of stations to be polled in SRMP is defined, the next process is to select the stations to be polled and to decide the polling sequence. At first, all stations are arranged in the order of ω_i , from low to high values. Since a lower value of ω_i indicates

that Station i has a higher probability of having frames to transmit, the stations with a lower value of ω_i are polled with higher priority. The next criterion that decides the polling sequence is E_i^k . For stations that have the same ω_i value, they will be arranged in an increasing order according to the E_i^k values. Since E_i^k represents the average number of transmitted frames during the previous W superframes, to achieve some of fairness, the stations with lower E_i^k values should have higher priority to be polled. From these two procedures, the polling sequence for all stations in BSS is obtained. The N_j stations in the polling sequence are polled by SRMP.

- **Step 3:** Prioritize different traffics

Different traffic types can be scheduled in the polling list according to their own priorities. After ordering the polling list, among the stations with the same ω_i and E_i^k values, the one with higher priority traffic should be assigned to the front of the list. For instance, if two stations have the same ω_i and E_i^k values, but have different traffic types, say, CBR and VBR. Assuming that CBR traffic has higher priority than VBR traffic, the station with CBR traffic will be put in the list ahead of that with VBR traffic.

- **Step 4:** Synchronize the polling time instants

We define the polling time instant as the time instant when a station is polled. When ω_i reaches one, Station i can be polled since it is at the beginning of the polling sequence. Therefore, the polling instant is closely related to ω_i . Since ω_i is estimated by the admission control unit, the polling instant in the time line is not synchronized with the actual time instant of frame generation. Consequently, we need to adjust the polling instant to minimize the delay. This process is called synchronization of polling instant.

We define a frame delay, T_d^i , to be the time duration from the time instant when one frame is generated in the MAC layer to the time instant when the frame is transmitted at Station i . When T_d^i is larger than T_{SF} , the PC changes the polling instant by updating ω_i as follows:

$$n_i = \begin{cases} SP_i - 1, & \text{if } T_d^i > \frac{(SP_i \cdot T_{SF})}{2} \text{ and } \omega_i = 1 \\ SP_i + 1, & \text{if } T_d^i \leq \frac{(SP_i \cdot T_{SF})}{2} \text{ and } \omega_i = 1 \\ \omega_i - 1, & \text{if } T_d^i > \frac{(SP_i \cdot T_{SF})}{2} \text{ and } \omega_i > 1 \\ \omega_i + 1, & \text{if } T_d^i \leq \frac{(SP_i \cdot T_{SF})}{2} \text{ and } \omega_i > 1 \end{cases} \quad (3-5)$$

The PC is not able to know T_d^i for each station, but each station knows T_d^i for its own frames. Thus, each station needs to inform its T_d^i to the PC when T_d^i larger than T_{SF} . For this purpose, the Subtype subfield in the frame control field of the MAC header is used by the uplink frames. Figure 3-7 shows the format of the frame control field. If the value of T_d^i is larger than $(SP_i \cdot T_{SF})/2$, the value of the Subtype subfield is set to 1000. Otherwise, the value of the Subtype subfield is set to 1001.

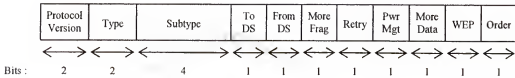


Figure 3-7. Format of Frame Control Field

3.3.3.2 Second scheduler for DTMP

From stations polled by SRMP, the PC obtains information such as the data rate in the physical-layer and the number of frames in the buffer at the MAC layer. According to

thes information, the PC allocates a TXOP to each station that has frames to transmit and with value of ω_i to be one. The TXOP for Station i is

$$TXOP_i = (T_{pre} + T_{PHY\_hdr} + T_{MAC\_hdr} + 2 \cdot T_{SIFS} + T_{ACK} + \frac{L_{Payload}}{R}) \cdot Q_i, \quad (3-6)$$

where T_{pre} , T_{PHY_hdr} , T_{MAC_hdr} , and T_{ACK} are the time durations of the preamble, the PHY header, the MAC header and the ACK frame, respectively. T_{SIFS} is the SIFS idle time.

$L_{Payload}$ is the length of the payload in bits, R is the data rate in the physical-layer, and Q_i is the number of frames in the buffer of Station i . There are two cases we should consider. The first is the case when the remaining time in the CFP after the DTMP frame is less than the sum of TXOPs of all stations with pending frames. The other is the opposite case. For the first case, the scheduler chooses the stations with nonzero Q_i values to be placed at the beginning of the polling list using ω_i and E_i^k as described in Section 3.2.1. Then, it sends the DTMP frame with the information, such as TXOP, Rate and AID of a selected station. For the second case, more stations are chosen from the polling sequence after N_j stations are chosen until the remaining time in the CFP is filled with their TXOPs. For stations not polled in SCP, but will be polled due to the second case, the Q_i value reported by the station in the previous superframe is used for TXOP allocation.

3.4. Simulation Setting

3.4.1 Wireless Channel Model

The wireless channel model is same as that adopted in Chapter 2 except the spread spectrum processing gain given by (2-6). For simplicity, we assume that each symbol is

chipped with an 11-chip pseudonoise (PN) code sequence regardless of modulation schemes. Therefore, the processing gain is 10.4 dB.

A minimum received power level for the carrier sensing is set to -95dBm, which is the noise power level. When the received power level is less than -95dBm, it is considered that the node can neither sense the channel nor demodulate the received frame. In this simulation, all stations are moving around with a slow pedestrian speed of 1m/s, within the coverage area of the BSS. Herein, it is assumed that the channel is constant during the period of one superframe.

Table 3-1. Simulation parameters for TS-MP performance evaluation

Parameter	Value
CWmin	31
CWmax	1023
SIFS time	10 us
DIFS time	50 us
Slot time	20 us
MAC header	272 bits
PHY header	48 bits
Preamble	144 us
ACK frame length	112 bits
RTS frame length	160 bits
CTS frame length	112 bits
<i>aCFPMaxDuration</i>	30 ms
Superframe duration	32 ms

The set of modulation schemes used in our simulation studies are BPSK, QPSK, 16QAM, 64QAM, and 256QAM. For simplicity, we ignore other common physical-layer components such as error correction coding. With 1MHz symbol rate and the above modulation schemes, the achieved data rates are 1, 2, 4, 6, and 8 Mbps, respectively. The relation between frame error rate (FER) and symbol error rate (SER) is given by (2-8) in

Chapter 2. We set the target FER to 8% according to the IEEE 802.11 standard [4]. The SER equation to determine the SNR are found [44]. For BPSK,

$$SER = Q\left(\sqrt{\frac{2E_s}{N_0}}\right), \quad (3-7)$$

and for QPSK and M-ary QAM,

$$SER \leq 1 - \left[1 - 2Q\left(\sqrt{\frac{3E_s}{(M-1)N_0}}\right)\right]^2, \quad (3-8)$$

where E_s / N_0 is the SNR per symbol and M is the signal constellation size. From the SER performance curves calculated from Equation 3-7 and Equation 3-8, the SNR ranges for the corresponding modulation schemes that the target SER is satisfied are given as follows, respectively,

$$R = \begin{cases} 1 \text{ (BPSK)} & , \quad SNR < SNR_2 \\ 2 \text{ (QPSK)} & , \quad SNR_2 \leq SNR < SNR_4 \\ 4 \text{ (16QAM)} & , \quad SNR_4 \leq SNR < SNR_6 \\ 6 \text{ (64QAM)} & , \quad SNR_6 \leq SNR < SNR_8 \\ 8 \text{ (256QAM)} & , \quad SNR_8 \leq SNR \end{cases} \quad (3-9)$$

where SNR_i is the SNR threshold for the data rate i to meet the target SER.

3.4.2 Network Setting

We assume that all stations except for the PC are uniformly distributed in the coverage area of an independent BSS with diameter 250 meters. The PC is always located at the center of the area. Since the proposed protocol and the other comparative protocols all have mechanisms to avoid collisions with stations in the neighboring BSS, we do not consider any neighboring BSS in this simulation. Moreover, only uplink traffic is considered. The synchronization problem with a preamble and the propagation delay

are not considered in our simulation. The parameters used in this simulation study are shown in Table 1. The choice of these parameters is based on the IEEE 802.11b DSSS standards. The duration of the CFP varies depending on the number of stations. If there is residual time in a CFP after all stations are polled or the PC broadcasts a CF-End frame, the residual time of CFP is merged with the CP. At least 4% of the superframe duration is assigned to the CP [4, 58]. Since the PCF is designed for the time-bounded services, we study two real-time traffic types, CBR and VBR, in the simulation. The traffic models for these traffic types are described as follows.

- CBR Voice Traffic Model

A voice source has two states, talkspurt and silence. Talkspurt is characterized by a voice activity detector (VAD) [66]. The durations of talkspurt and silence are exponentially distributed with mean values of t_1 and t_2 , respectively. The values of t_1 and t_2 are set to 1.0 and 1.35 seconds, respectively. We use a 16 kbps voice traffic source to generate one 200 bytes payload voice frame every 0.1 seconds during the talkspurt period. We assign the delay time limit of a voice frame to 0.1 seconds. That is, all voice frames must be transmitted before the next frame arrives.

- VBR MPEG-4 Traffic Model

In our simulation, the trace statistics of actual MPRG-4 video streams reported [67-68] are used. We use the video stream of Star Wars IV, which has a mean bit rate of 53 Kbps and a peak rate of 940 Kbps. The size of video packet is set to 800 bytes based on a study of Duel-Hallen et al.[61]. According to the mean bit rate of 53 Kbps, one video packet is generated every 0.12 seconds. Herein, the delay limit of video packet is set to 0.12 seconds, so that all video frames must be transmitted before the next frame arrives.

Each station has either one CBR or one VBR flow. In this simulation, three measurements for performance evaluation are considered: dropping probability, average delay, and CFP throughput. The average delay is defined as the time duration from the arrival of a frame in the MAC layer to the departure of the frame. It is assumed that the instant that a frame is generated is the same as that of the frame arrival in the MAC layer. The CFP throughput is

$$Throughpt_{CFP} = \frac{\sum_{j=1}^{N_{SF}} \sum_{i=1}^{N_{STA}} Data_i^j}{\sum_{j=1}^{N_{SF}} T_{CFP}^j}, \quad (3-10)$$

where T_{CFP}^j , N_{SF} , N_{STA} , and $Data_i^j$ are the used CFP duration in j th superframe, the total number of superframes, the number of stations and the transmitted data bits at station i in the j th superframe, respectively.

We simulate 200 different realizations with different positions of stations. Each scenario is simulated for 60 seconds. In every realization, the channel condition for each communication link is recalculated according to the distance between any two stations and the shadowing environment for each station.

3.5. Performance Evaluation

3.5.1 Performance Comparison with Round-Robin Scheduling Scheme

In Section 3.5, we compare the performance of two existing protocols with that of our proposed protocol. The first protocol is the contention free single polling (CF-SP) scheme with RTS/CTS frames in 802.11e and the second protocol is CP-MP. To evaluate the efficiencies of these two protocols and simplify the simulations, a Round-Robin scheduling scheme is used for all protocols. For the same purpose, it is assumed that the

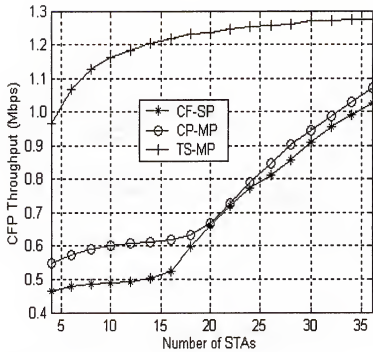


Figure 3-8. Average CFP throughputs

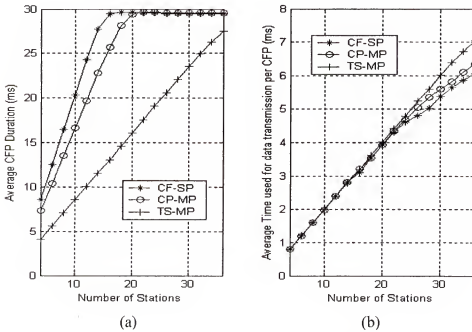


Figure 3-9. Other performance evaluations. A) Average CFP duration. B) Average time used for data transmissions.

channel is constant during a simulation of one realization, and the physical transmission rate of each communication link is selected via the rate decision process described in Section 3.4.1. These assumptions for the channel and the rate are also applied to Section 3.4.4.

Figure 3-8 shows the CFP throughputs of the three protocols when stations with CBR traffic and stations with VBR traffic co-exist in the BSS. The number of stations with CBR traffic is the same as that with VBR traffic. The performance of TS-MP is 18% to 140% higher than that of CF-SP and 13% to 100% higher than that of CP-MP. However, we observe that the CFP throughputs of CF-SP and CP-MP increase rapidly after a certain number of stations, 18 for CF-SP and 22 for CP-MP. This is elucidated through the analyses of Figs. 3-9(a) and (b). Figure 3-9(a) shows the average CFP duration in a simulation, and Figure 3-9(b) shows the average time used for data transmissions, which is the MAC payload, in a CFP. The average CFP duration increases rapidly in CF-SP and CP-MP comparing to that in TS-MP. In addition, it is saturated at 18 stations for CF-SP and 22 for CP-MP since the maximum CFP duration is constrained in Table 1. However, in Figure 3-9(b), the results for the time used for the data transmission are not distinguishable for three protocols. This indicates that CF-SP and CP-MP require more time for serving the same number of stations than TS-MP does. That is, the time consumed by the overheads in CF-SP and CP-MP in the CFP is much more than that in TS-MP as described in Section 3.3. Therefore, the CFP duration in CF-SP and CP-MP reaches $aCFPMaxDuration$ earlier than that in TS-MP. The rapid increase in the CFP throughputs for CF-SP and CP-MP is elucidated by the saturation of the CFP. Figures 3-10(a) and (c) show the dropping probabilities, and Figs. 3-10(b) and d show the

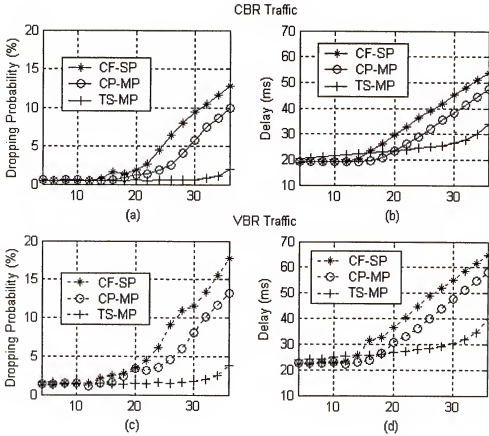


Figure 3-10. Dropping probability and average delay as functions of the number of stations. A) Dropping rate of CBR traffic. B) Delay of CBR traffic. C) Dropping rate of VBR traffic. D) Delay of VBR traffic.

average frame delays for the CBR and VBR traffic. As the number of stations increases, the dropping probabilities of TS-MP reduces up to 87 % of that of CF-SP and 80% of that of CP-MP for both traffic types. However, we observe that the average delay of TS-MP with a small number of stations is larger than those of the other protocols. This is caused by SCP in TS-MP. In each CFP, the first data frame in TS-MP is transmitted after the transmission of the DTMP frame. That is, most of the overhead in TS-MP is placed in the front of the CFP, whilst the overhead is distributed to each data frame transmission in CF-SP and CP-MP. Therefore, when the number of stations is small, the delay due to the overhead of the TS-MP protocol appears prominently. However, as the number of

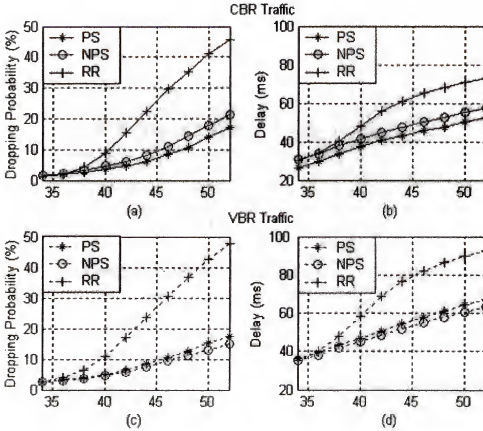


Figure 3-11. Dropping probabilities and average frame delays of the three configurations. A) Dropping rate of CBR traffic. B) Delay of CBR traffic. C) Dropping rate of VBR traffic. D) Delay of VBR traffic.

stations increases, all stations in CF-SP and CP-MP cannot be served during current CFP because the required time to serve all stations passes over the maximum CFP duration as explained previously with Figure 3-9. Thus, some of the stations are polled in the next CFP, which causes an additional delay. On the other hand, in TS-MP, most of stations are served in the current CFP so that the delay increases slowly as shown in Figs. 3-10(c) and (d).

3.5.2 Performance Evaluation with the Proposed Scheduling Scheme

In Section 3.5.1, simulation results show that the proposed protocol provides better performance than the other two protocols. Now, we evaluate the performance of the

proposed protocol with the proposed scheduling scheme. In Section 3.5.2, three configurations are compared:

- **Case 1:** TS-MP with Round Robin scheduling (RR);
- **Case 2:** TS-MP with the proposed non-priority-based scheduling (NPS); and
- **Case 3:** TS-MP with the proposed priority-based scheduling (PS).

In our simulation, the CBR traffic has higher priority than the VBR traffic. The number of stations in the BSS increases from 34 to 52 with a step size 2, and the number of stations with CBR traffic is the same as the number of stations with VBR traffic.

As shown in Figure 3-11, the frame dropping probability of case 2 for CBR traffic and VBR traffic reduces up to 67% comparing to that of case 1. The average time delay of case 2 improves 28% for CBR traffic and 32% for VBR traffic comparing to that of case 1. With a Round-Robin scheduling scheme, the PC has to poll all stations in SCP so that the CFP can be dominated by the two polling frames from the PC and the SR frames from stations when the number of stations are large. On the other hand, by dynamically changing the number of stations in the polling sequence of SCMP based on the expected buffer status for each station, the proposed scheduling scheme not only prevents the CFP from being dominated by the two polling frames and the SR frames, but also increases the time portion for data transmission. These results are reflected in the CFP throughput as shown in Figure 3-12. Using the proposed scheduling scheme, performance improvement is achieved through removing unnecessary overhead.

Now, we compare the performance of case 3, under which stations in the BSS are scheduled with different priorities depending on the traffic type. The dropping probability and the average frame delay of the CBR traffic in case 3 decreases up to 46% and 21%

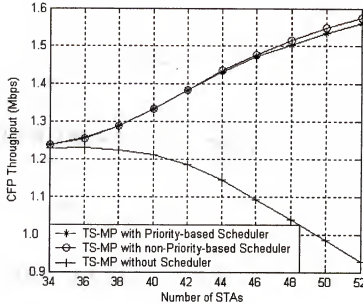


Figure 3-12. Comparison of CFP throughputs for the three configurations compared to corresponding values in case 2. For the VBR traffic, the dropping probability and the average frame delay in case 3 are up to 7% and 20% higher than the corresponding values in case 2. These results reflect the fact that the CBR traffic is given higher priority by the scheduler.

3.5.3 Rate-Adaptation (RA) Functionality

Now, we show the adaptability of our protocol over the time-varying channel. The protocols are evaluated under the Ricean fading channel with Ricean parameter, K , set to 5. Thus, all communication links experience different channel condition on every superframe. Figure 3-13 shows the frame dropping probabilities of TSMP with RA and one without RA. Using the rate adaptation function of TSMP, we can reduce the dropping probability by 70% up to 98% comparing to that of TSMP without RA. This shows the adaptability of our polling scheme against the time varying wireless channel.

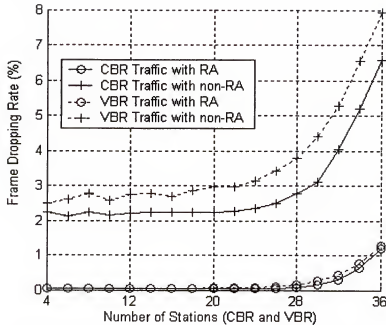


Figure 3-13. Conventional fragmentation process and the timeline of data transmission with rate adaptation

3.6. Conclusion

In this chapter, we propose a new polling-based MAC protocol for the PCF in IEEE 802.11 WLAN. The major innovation is the use of two multipolling frames with different purposes. Through the first multipolling, the PC obtains information required to schedule the polling sequence for data transmission. The second multipolling coordinates data transmissions without collision. Comparing with the single polling scheme used in the conventional IEEE802.11 MAC protocol, the proposed scheme can reduce the overhead caused by the polling frames. For most previously proposed protocols in the literatures, the PC does not have information about the appropriate physical transmission rate for communication link and the buffer status of each station involved in the PCF. As a consequence, simple scheduling schemes, such as the Round-Robin scheduling scheme, is used. However, the proposed protocol makes it possible for the PC to schedule the

polling sequence based on the currently obtained information from all stations. Therefore, by utilizing the information, we can design more efficient scheduling schemes. From the extensive performance simulation, we have shown that the proposed polling-based MAC protocol gives significant performance improvements over the other polling-based MAC protocols.

CHAPTER 4 FEEDBACK-ASSISTED MAC PROTOCOL FOR REAL-TIME TRAFFIC IN HIGH RATE WIRELESS AREA NETWORKS

4.1. Introduction

Wireless connectivity has revolutionized consumer electronics and personal computer peripherals. High performance wireless networking solutions are replacing today's wired devices such as USB and 1394 due to their greater flexibility and simpler installation requirements. In addition, supported by emerging standards such as 802.11 and 802.15, it is anticipated that wireless technology will eventually be used to replace the tangle of wires needed to transfer video and audio signals.

As mentioned in Chapter 1, WPANs providing from 5 to 50 meters' range wireless connectivity have been studied by the IEEE 802.15 WG. The first standard of the IEEE 802.15 WG is IEEE 802.15.1 [74], which is a Bluetooth-based technology. The features of this technology are low power consumption, low data rate, low cost and small package size. The data rate of Bluetooth is up to 1 Mbps.

The next generation technology of WPAN targets consumer electronics and portable communication devices that require higher data rates. The IEEE 802.15.3 TG has been chartered to create a HR-WPAN standard and has recently published a final standard [8].

The target applications of HR-WPAN can be divided into two categories. The first application is multi-megabyte data file transfers such those involving image and music files. The second application is distribution of real-time video and high-quality audio,

which are strictly time-bounded applications. To support higher data rates and better QoS, HR-WPAN adopts a Time Division Multiple Access (TDMA)-based MAC protocol that will be described in Section 4.2. In HR-WPAN, a pair of nodes can communicate through peer-to-peer connectivity without contention during an allocated channel time. As the quality of video and audio improves, the amount of data required to be delivered between consumer electronics also increases, so higher speed wireless connectivity is required. At this point, since the Federal Communications Commission (FCC) has approved the commercial use of UWB technology [9], the IEEE 802.15.3a SG was established to study UWB technology for use in the physical (PHY) layer in HR-WPAN. Using UWB technology, the maximum achievable data rate could be around 500 Mbps, as suggested by Barta [75]. Combined with the high data rate of UWB, the multimedia-oriented features of the 802.15.3 medium access controller provide the QoS provisions needed for streaming HDTV and other multimedia applications.

Although the MAC protocol in the IEEE 802.15.3 standard is expected to play a crucial role in the formation of home networks or small office networks, significant efforts to improve the performance of the MAC protocol have not been made since the standard was published recently. Performance enhancements undertaken by informing queue status (Q-status) of each node to a piconet controller (PNC) are shown in the proposal of Mangharam et al. [40]. In this scheme, the number of pending packets at each DEV is included in the MAC header of every packet. Thus, by overhearing every packet exchange, a PNC can allocate appropriate channel time for transmitting packets stored at a DEV in the next superframe. This scheme aims to handle VBR traffics and adopts a flexible superframe size. One potential drawback is that the size of the superframe may

change too frequently. This may introduce some difficulties in timing accuracy and positioning for strictly time-bounded applications, as suggested [76, 41]. Furthermore, the piggybacked information can be useful only when there is a burst to transmit. Moreover, the channel time allocation algorithm for different traffic types is not considered. An algorithm proposed [41] focuses on utilizing wasted or remaining channel times and uses a constant superframe size. A superframe with two static channel times is used: one for CBR traffic and the other for real-time VBR (rt-VBR) traffic. Also, this scheme does not consider how to allocate the channel times.

Rhee et al. [42] propose a channel time allocation scheme for a specific application, MPEG 4 traffic. Since packets generated from a MPEG 4 encoder are classified into three types and are arranged in a periodic pattern, a central device can allocate channel time for transmissions of MPEG 4 packets according to the packet pattern. A packet transmission method without a preamble is introduced by Brabenac [43] because the physical preamble overhead remains as a dominant factor to be overcome in the high transmission rate UWB technology. A rate-adaptive MAC protocol for HR-WPAN is proposed [44]. Based on the estimated channel quality through using the received packet, the receiver chooses an appropriate data rate and sends it back to the transmitter. The target applications considered [44] have an asynchronous, bursty data transmission requiring an acknowledgement feedback such as MP3 file transfer. However, this method is not applicable for real-time services, which do not require acknowledgement feedback.

In this chapter, we propose an enhanced MAC protocol for HR-WPAN to support strictly time-bounded services more efficiently and to adapt the physical transmission rate according to the time-varying channel condition. In Section 4.2.1, the MAC protocol in

the IEEE 802.15.3 standard is briefly described. In addition, the way to support multi-rates as defined in the standard is illustrated in Section 4.2.2.

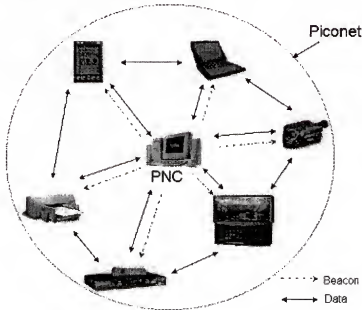


Figure 4-1. A piconet in IEEE 802.15.3

4.2. High-Rate Wireless Personal Area Network in IEEE 802.15.3

4.2.1 MAC Protocol

In the HR-WPAN standard specifications, DEVs are communicating on a centralized and connection-oriented ad-hoc network called piconet as shown in Figure 4-1. One of the participating DEVs must be designated as a piconet coordinator (PNC). The PNC provides basic timing information for the operation of the piconet and manages the quality of service (QoS) for delay sensitive applications.

The MAC layer in the IEEE 802.15.3 standard employs a time-slotted superframe structure. Figure 4-2 illustrates the superframe structure in the HR-WPAN standard. The superframe consists of three major parts: a beacon, an optional Contention Access Period (CAP) and a Channel Time Allocation Period (CTAP). The beacon packet is transmitted by the PNC at the beginning of each superframe. It allows all DEVs in a piconet know

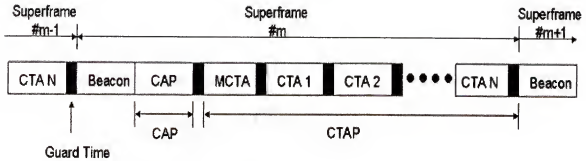


Figure 4-2. Superframe structure of IEEE 802.15.3

about the specific information for controlling a piconet, such as superframe duration, channel time allocations, used frequencies and etc. The CAP is used for transmissions of short and non-QoS data packets and command/response packets. The medium access mechanism during the CAP is Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA). The remaining period in the superframe is CTAP. The CTAP is composed of Channel Time Allocation (CTA) periods and Management Channel Time Allocation (MCTA). While MCTA like CAP is used for sending command packets, the slotted ALOHA mechanism is used for channel access. When a DEV needs a CTA on a regular basis, it sends a channel time request (CTRq) command to the PNC during the CAP or MCTA. Thus the PNC decides the duration of the superframe, CAP, and CTAP based on

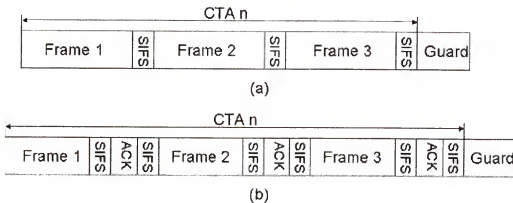


Figure 4-3. Packet transmissions. A) with No-ACK. B) Imm-ACK in a CTA.

the DEVs' requests. During one CTA period, one DEV can transmit several packets to one target DEV without collision. Each packet transmission may be followed by an acknowledgement (ACK) packet. A Short InterFrame Spacing (SIFS) idle time is added for a sufficient turnaround time between two consecutive packet transmissions in a CTA. In addition to SIFS, a guard time is required to prevent collision of two adjacent CTAs. Although the scheduling algorithm for allocating CAP, MCTAs, and CTAs plays a critical role on a performance of WPAN, such algorithm is not specified in the 802.15.3 standard.

The specification for the MAC protocol defines three acknowledgement types: no-acknowledgement (No-ACK), immediate-acknowledgement (Imm-ACK) and delayed-acknowledgement (Dly-ACK). An Imm-ACK is transmitted from the destination DEV when a transmitted packet is received correctly, while in the No-ACK case, no ACK is transmitted to the source DEV. A Dly-ACK is used only for directed stream data packets (e.g., isochronous connection). Figure 4-3 illustrates packet transmissions with No-ACK and Imm-ACK in a CTA.

4.2.2 Multi-Rate Support

The IEEE 802.15.3 physical (PHY) layer is operating in the unlicensed frequency band between 2.4 GHz and 2.4835 GHz. The symbol rate is 11 Mbaud. The raw PHY layer data rates are 11 Mbps for uncoded QPSK modulation, and 22, 33, 44, and 55 Mbps for trellis-coded QPSK, 16/32/64-QAM, respectively. The specification in the IEEE 802.15.3 MAC suggests two methods to obtain channel condition information and to select the data rate for transmission. The first method is to periodically transmit the channel status request command to the target DEV. When receiving that command, the target DEV sends a channel status response command back to the transmitting DEV. The

channel status response command includes the number of successfully received packets, the number of erroneous packets and the number of measured packets. The source DEV decides the data rate based on this information. In the second method, the channel condition is evaluated by the presence or absence of ACKs for the transmitted packets. This information is used to decide the data rate for the next packet transmissions. However, the second method is not applicable for the case of using No-ACK. If the Dly-ACK mechanism is used, all packets in a burst are transmitted with the same data rate.

4.3. Proposed MAC protocol

4.3.1 Motivation

Even small delay in HR-WPAN may cause serious performance degradations since HR-WPAN is targeting on delay-constrained real time multimedia services with a bulky traffic size and high bit rate, such as home theater systems with HDTV. Therefore, the channel time allocation algorithm plays an essential role to guarantee delay bound performance of real-time applications in HR-WPAN. Nevertheless, it is not proposed in the previous works. Furthermore, the information delivered by a CTRq command as

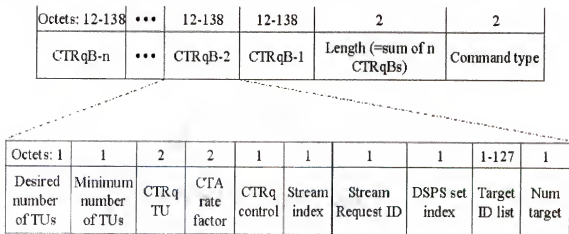


Figure 4-4. Channel time request command format and channel time request block field format

shown in Figure 4-4 is insufficient for the PNC to decide the duration and location of a CTA for the requesting DEV. The IEEE 802.15.3 TG considers the scenario that DEVs frequently join and leave a piconet as mentioned by Gandolfo et al. [7]. In this scenario, many factors, such as a superframe length and a number of flows, vary in time. As a consequence, the CTA allocation algorithm is required to support the QoS requirements over these variable factors.

In wireless networks, channel conditions need to be estimated to dynamically choose the appropriate transmission data rate over the time varying wireless channel, so that the higher performance can be achieved. As illustrated in Section 4.2.2, the channel condition in IEEE 802.15.3 is estimated based on the results of attempted transfers of data packets between two DEVs that are actively participating in a data transfer. However, using this method can not cope with fast channel changes and may cause incorrect channel information which leads to performance degradation. Moreover, for traffics with long packet inter-arrival time, this estimation method are futile since the transmission history for such a long time period can not represent the current channel condition. Recently the use of Signal-to-Noise Ratio (SNR) has been suggested to estimate the channel condition. The two methods [21], using transmission history and SNR, for the channel condition estimation are evaluated over a WLAN environment. The evaluation [21] shows that the method using SNR achieves a higher performance gain than that using the result of attempted transfers of data packets. However, this formal method requires feedback information from the receiver, which is not applicable to real time applications without acknowledgements.

With these considerations, we propose an enhanced MAC protocol for time-bounded services in Section 4.3.2.

4.3.2 Proposed Protocol for High-Rate Wireless PAN

4.3.2.1 Channel time allocation algorithm

As mentioned in Section 4.3.1, providing delay-bounded services is critical to the real-time traffics and no algorithm to allocate channel times is specified in the standard. Here, we propose a channel time allocation algorithm to synchronize a CTA to the packet arrival instant. We introduce two main parameters that affect the channel time allocation process. The first one is the service period of DEV i , LA_i . The value of LA_i is the estimated inter-arrival time of packets at DEV i with payload P_i . It is given by

$$LA_i = \left\lceil \frac{P_i \cdot 8}{M_i} \right\rceil, \quad (4-1)$$

where P_i and M_i are the payload in the MAC packet in bytes and the data arrival rate for CBR traffic (or the mean arrival rate for real-time VBR (rt-VBR) traffic) in the MAC layer at DEV i , respectively. LA_i is calculated by DEV i and informed to the PNC using the channel time request command. In a general wireless network, the two parameters, P_i and M_i , can be obtained from the admission control unit in the central controller during the association period [5, 77]. For this purpose, the channel time request command shown in Figure 4-4 is modified. The CTA rate factor field in the channel time request command is changed to the Traffic arrival rate field. We define another parameter Ptr_i , which is related to LA_i in order to allocate CTA for DEV i . Ptr_i is a timer which is initialized to be LA_i and decreased as time elapse. The moment when Ptr_i reaches zero is

the time instant to allocate CTA for DEV i. That means that the Ptr_i indicates the remaining time for the CTA allocation for DEV i.

At first, the PNC gathers DEVs whose Ptr_i s are less than the current superframe duration since CTAs of those DEVs must be allocated in the current superframe. Therefore, the ensuing steps are applicable only to those DEVs. Then, the PNC decides the number of CTAs which will be allocated in the current superframe. The PNC needs information of $NumCTA_i$, ST_i^j and DT_i^j for each DEV to allocate CTAs in the superframe. $NumCTA_i$ is the required CTAs for a DEV i during a superframe period. It is defined as

$$NumCTA_i = \left\lfloor \frac{T_{SF} - Ptr_i}{LA_i} \right\rfloor + 1, \quad (4-2)$$

where T_{SF} is the time duration of the superframe. ST_i^j is the time instant of the beginning of CTA j for DEV i. It is defined as

$$ST_i^j = Ptr_i + (j-1) \times LA_i, \quad 1 \leq j \leq NumCTA_i. \quad (4-3)$$

Note that ST_i^j is less than T_{SF} . DT_i^j is the time duration to be allocated to CTA j of DEV i. It is calculated as

$$DT_i^j = \left[T_{OH} + T_{SIFS} + \frac{(L_{len}^j + L_{FCS})}{R_i} \right] \cdot Q_i + T_{guard}, \quad (4-4)$$

where T_{OH} is the time overhead including the preamble, PHY header, MAC header, Header Check Sequence (HCS), and guard time. In the IEEE 802.15.3 standard, the value of T_{OH} at 11 Mbps is different from those at the other rates. T_{SIFS} is the SIFS idle time.

L_{len}^i is the length of the payload in bits for DEV i. L_{FCS} is the length of the frame check

sequence (FCS). R_i is the data rate in the physical-layer and Q_i is the number of packets to be transmitted during CTA j of DEV i . The beacon packet in a superframe has information fields for the location and duration of all CTAs as described in the IEEE 802.15.3 standard. Thus, the proposed scheme can be implemented without any additional modification to the standard.

Now, CTAs are allocated at time ST_i^j with duration DT_i^j on a superframe. When several CTAs overlap, the CTA with lower ST_i^j is allocated in advance of the one with higher ST_i^j . However, the CTAs can also be allocated based on same specified performance requirements such as priority and throughput. In the former case, CTAs of DEV with higher priority are allocated ahead of those from another DEV with lower priority. In the latter case, CTAs of a DEV with a higher transmission data rate is allocated ahead of one with lower data rate. If there is time remaining between two consecutive CTAs, this duration becomes MCTA for transmitting command packets. However, if the remaining time is less than the threshold T_{thr} , it is merged to previous or next CTA. Therefore, MCTA allocation is also defined. The threshold T_{thr} is a sum of the slot time and the time duration of a CTRq packet. This choice ensures that at least one command packet can be transmitted in the MCTA. The total duration of CTAs and MCTAs allocated in a superframe should be less than T_{SF} . If its total duration is larger than T_{SF} , CTAs at the end will be removed until it is less than T_{SF} .

At the final step, P_{tr_i} is reset to a value for the next superframe formation. This value is given by

$$P_{tr_i} = LA_i - (T_{SF} - ST_i^{last}), \quad (4-5)$$

where ST_i^{last} is the time duration of the lastly allocated CTA for DEV i . For a DEV whose CTAs are not allocated in this superframe, the corresponding Ptr_i is subtracted by T_{SF} .

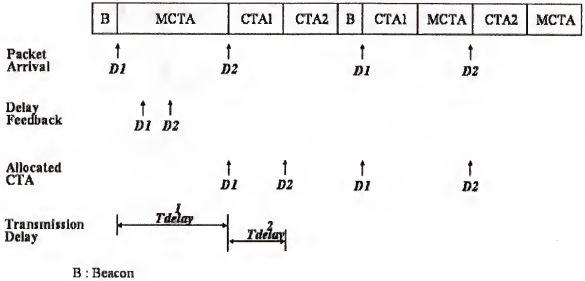


Figure 4-5. An example of CTA synchronization

4.3.2.2 Feedback-assisted CTA allocation

Employing CTA allocation algorithms based only on statistical packet inter-arrival time is not sufficient to overcome the aforementioned problem for strictly time-bounded services. Since information given by a channel time request command does not include the optimal time instant of a CTA, the PNC may allocate the CTA at any position within a superframe. This causes time wasted from packet arrival at the MAC layer to the transmission of that packet. This wasted time is called transmission delay. Figure 4-5 shows an example of transmission delay caused by the lack of information about the actual packet arrival instant at the PNC. This delay increases as the packet inter-arrival time increases and may maintain until the end of the flow. Furthermore, it can be longer in heavy load cases since several CTAs overlap. Because of this problem, rt-VBR traffics

Octets: 10	1	1-4	2	4
MAC header	Report ID	Report Payload	Length	FCS

Figure 4-6. Status report command packet format

whose packet inter-arrival time is variable cannot be handled. For rt-VBR traffic, instantaneous bit rate fluctuates widely about a mean value [72, 73]. As a consequence, the inter-arrival time at DEV i also fluctuates and is different from LA_i , statistically calculated by the PNC. That means that more than one packet can be stored in the buffer at the instant CTA allocation. If PNC allocates CTAs for rt-VBR traffic using the peak inter-arrival time, utilization of channel time will be degraded.

To overcome these problems, we propose a feedback-assisted CTA allocation method. To achieve better CTA allocation, each DEV informs its current status to the PNC. For this purpose, during the MCTA, a DEV sends the status information to the PNC by using the status report command packet shown in Figure 4-6. This command packet specifies three statuses of a DEV: Q-status, delay, and physical transmission rate. The Report ID subfield in the status report command indicates one of seven possible report types and the Report Payload subfield is the value of each reporting item. Table 1

Table 4-1. List of report IDs and report payload sizes

Report Type	Report ID	Report Payload Size (Octet)
Q-Status	0001	1
Delay	0010	2
Rate	0011	1
Q-Status + Delay	0100	3 (1+2)
Q-Status + Rate	0101	2 (1+1)
Delay + Rate	0111	3 (2+1)
Q-Status + Delay + Rate	1000	4 (1+2+1)

lists the Report ID and the size of Report Payload. When the PNC receives a status report command with the delay information from DEV i , the value of Ptr_i of DEV i at the PNC is subtracted by that delay. Hence, a CTA for DEV i in the next superframe will be allocated earlier than the current CTA position since Ptr_i is shortened by the status report command.

Figure 4-5 illustrates an example of the CTA synchronization process with packet arrivals. In the first superframe, DEVs D1 and D2 have the transmission delays, T_{delay}^1 and T_{delay}^2 , respectively. The transmission delay information is sent during the MCTA of the first superframe. The PNC changes the time instant of the CTAs in the second superframe. Thus, from the second superframe on, CTAs are located at the packet arrival time instants and the transmission delay becomes zero. If the packet arrival rate is constant as CBR traffic, a single status report with delay information is enough for the PNC scheduler since it a DEV with CBR traffic generates one packet in each inter-arrival time. However, for rt-VBR traffic, this assumption is not guaranteed as mentioned before. In order to dynamically allocate the duration of CTAs for DEVs with rt-VBR, the queue status of each DEV needs to be reported to the PNC scheduler frequently. This queue status information is also transmitted using the status report command during the MCTA. This information is used in Equation 4-4 to provide the value for the parameter Q_i .

We use channel estimation information from the physical-layer at a receiver to choose the transmission data rate. A rate adaptation mechanism for best effort traffic types such as the bulk file transfer [44] is proposed. On the other hand, since we are dealing with time-bounded real-time services with No-ACK policy here, a packet to inform the data rate to the sender is needed. For this purpose, the aforementioned Status

Report command is used to report the selected data rate to the PNC as well as the sender. This command is transmitted during a CAP or MCTA only when the currently used rate is not appropriate to meet certain performance criteria like the Packet Error Rate (PER). The channel estimation process is done by the physical-layer. This feedback rate information is utilized for decision of the CTA durations in the next superframe as shown in Equation 4-4.

In the proposed scheme, the transmission of status report commands plays an important role in allocating CTAs in a superframe. However, the PNC may form a superframe without any MCTA due to a heavy traffic load or an insufficient superframe size. To ensure at least one status report command can be transmitted in a superframe, the PNC allocates at least one MCTA with the minimum MCTA time duration. Moreover, the last channel time in a superframe must be a MCTA, called Essential MCTA (E-MCTA). This allows the latest status information of each DEV to be delivered to the PNC and reflected in the next superframe.

4.4. Performance Analysis

4.4.1 Networking Setting

We assume that all DEVs except the PNC are uniformly distributed in the coverage area of a piconet with diameter 20 meters. The PNC is always located at the center of the area. We do not consider any neighboring piconet in this simulation. Moreover, perfect synchronization in the physical-layer is assumed and the propagation delay is not considered. The parameters used in this simulation study are shown in Table 2. The choice of these parameters is based on the IEEE 802.15.3 standards [8].

Since the proposed scheme is designed for the time-bounded services, we study two real-time traffic types, CBR and real rt-VBR in the simulation. The CBR traffic flow is

Table 4-2. Simulation parameters based on IEEE 802.15.3 standard

Parameter	Value
SIFS time	10 us
Guard time	50 us
Slot time	59 us
MAC header	10 octets
PHY header	2 octets
Preamble	17.5 us
HCS	16 bits
FCS	32 bits
Minimum MCTA	3 ms

generated at 912 kb/s. This rate is the maximum bit rate of the MPEG audio encoder [78].

For the rt-VBR traffic model, the trace statistics of actual MPRG-4 video streams reported by Fitzek et al. [72, 73] are used. We use a high quality video stream from “Silence of the Lambs”, which has a mean bit rate of 580 Kbps and a peak rate of 4.4 Mbps. Each DEV has either a CBR or rt-VBR traffic flow. A DEV alternates between the two states, ON and OFF, and their durations are exponentially distributed with mean values of 20.0 sec and 0.05 sec, respectively. A traffic flow is generated only during ON state. At the beginning of the ON state, a DEV selects a destination DEV and transmits a CTRq command to the PNC during a MCTA. In this simulation, CAP allocation is not considered since it is optional in the standard [8]. In addition, three measurements for performance evaluation are considered: Job Failure Ratio (JFR), average transmission delay, and PER. The JFR is the packet dropping rate because of missing delay bound [40, 41]. The average transmission delay is defined as the time duration from the arrival of a packet in the MAC layer to the departure of the packet or dropping of it. It is assumed that a packet arrives at the MAC layer at the instant that it is generated.

The scheme proposed in this chapter, namely Feedback-Assisted WPAN (FA-WPAN), is compared with the HR-WPAN scheme suggested by Mangharam et al. [40]. HR-WPAN adopts an aggressive CTA allocation algorithm. CTA durations for both CBR and rt-VBR traffic flows are evenly allocated over the superframe duration in the allocation algorithm [40]. However, since the rt-VBR traffic may generate more packets than the CBR traffic does, it is unfair to allocate same CTA durations for both traffics. Therefore, in this simulation, the CTA duration for the rt-VBR traffic is roughly two-time longer than that for the CBR traffic. HR-WPAN also allocates a MCTA of 3 ms duration as the first CTA in every superframe. Therefore, the duration of each CTA is

$$\frac{(T_{SF} - T_{bec} - T_{MCTA})}{(1 \cdot N_{vbr} + 2 \cdot N_{cbr})} \times \begin{cases} 2 & \text{for rt-VBR} \\ 1 & \text{for CBR} \end{cases},$$

where T_{bec} and T_{MCTA} are time durations of the beacon packet and E-MCTA, respectively. N_{vbr} and N_{cbr} are the number of flows of rt-VBR and CBR traffics, respectively. The position of the MCTA in HR-WPAN does not affect to the performance since no command packet, except the CTRq command, is considered.

Each scenario is simulated for 10 minutes. For evaluation of the rate adaptation scheme, we simulate 50 different realizations with different positions of DEVs. In every realization, the channel condition for each communication link is recalculated according to the distance between any two DEVs.

4.4.2 Wireless Channel Model

We employ the log-distance path loss channel model [57]. The path loss $\overline{PL}$ at distance d is

$$\overline{PL}(d)[dB] = \overline{PL}(d_0)[dB] + 10n \log\left(\frac{d}{d_0}\right), \quad (4-6)$$

where d_0 is the close-in reference distance and n is the path loss exponent. We set n to 3.3 according to the SG3a alternate PHY selection criteria [79]. To estimate $\overline{PL}(d_0)$, we use the Friis free space equation

$$P_r(d_0) = \frac{P_t G_t G_r \lambda^2}{(4\pi)^2 d_0^2 L}, \quad (4-7)$$

where P_t and P_r are the transmit and receive power, G_t and G_r are the antenna gains of the transmitter and receiver, λ is the carrier wavelength, and L is the system loss factor which is set to 1 in our simulation. The transmit power and antenna gain are set to 0 dBm and 0 dBi [79], respectively. The received power is

$$P_r(d)[dBm] = P_t[dBm] - \overline{PL}(d). \quad (4-8)$$

Finally, the long-term signal-to-noise ratio is

$$SNR_L[dB] = P_t - \overline{PL}(d) - N, \quad (4-9)$$

where N is the noise power set to -95 dBm.

To demonstrate the functionality of the rate adaptation scheme in our proposed protocol, the received SNR_L is varied by the Ricean fading gain α , which is generated according to the modified Clarke and Gans fading model [60]. Under this model, the SNR of the received signal is

$$SNR[dB] = 20 \cdot \log_{10} \alpha + SNR_L[dB]. \quad (4-10)$$

For the data rate in the physical-layer for each communication link, we assume that the system adapts the data rate by properly choosing one from a set of modulation schemes according to the channel condition. The set of modulation schemes used in our simulation studies are BPSK, QPSK, 8QAM, 16QAM, and 32QAM. For simplicity, we ignore other

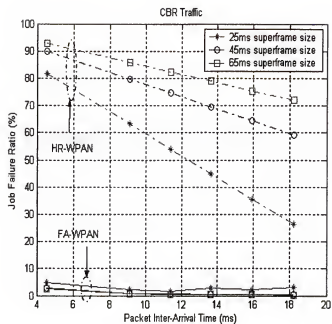
common physical-layer components such as error correction coding. With 11MHz symbol rate and the above modulation schemes, the achieved data rates are 11, 22, 33, 44, and 55 Mbps, respectively, which are same data rates in the standard. The relation between Packet Error Rate (PER) and Symbol Error Rate (SER) is given by (2-8) in Chapter 2. We set the target FER to 8% according to the IEEE 802.15.3 standard [8]. The SER equations for different modulation schemes to determine the SNR are given by (3-7) and (3-8) in Chapter 2. The SNR ranges for the corresponding modulation schemes that the target SER is satisfied are given as follows, respectively,

$$R = \begin{cases} 11 \text{ (BPSK)} & , \quad SNR < SNR_{22} \\ 22 \text{ (QPSK)} & , \quad SNR_{22} \leq SNR < SNR_{33} \\ 33 \text{ (8QAM)} & , \quad SNR_{33} \leq SNR < SNR_{44} \\ 44 \text{ (16QAM)} & , \quad SNR_{44} \leq SNR < SNR_{55} \\ 55 \text{ (32QAM)} & , \quad SNR_{55} \leq SNR \end{cases} \quad (4-11)$$

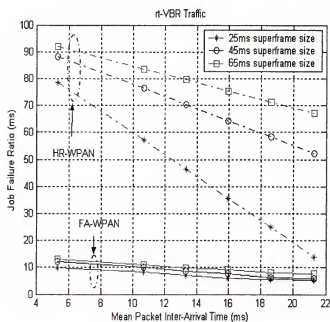
where SNR_i is the SNR threshold for the data rate i to meet the target SER.

4.4.3 Performance Evaluation

In Section 4.4.3, the proposed protocol is evaluated with three superframe sizes, 25, 45, 65 ms, over an error free wireless channel. The maximum superframe size described in the IEEE 802.15.3 standard is 65536 μ s. The delay bound is set to the packet inter-arrival time [34, 80]. That is, all packets arriving at the MAC layer must be transmitted before the next packet arrives. The inter-arrival times are varied by changing the packet size with a constant traffic bit rate for CBR (or the mean traffic bit rate for rt-VBR). The packet sizes used in this simulation are 512, 1024, 1286, 1536, 1792, and 2048 octets defined as preferred packet sizes in the IEEE 802.15.3 standard.

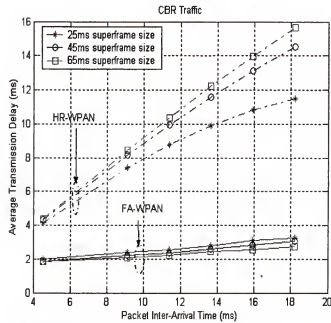


(a)

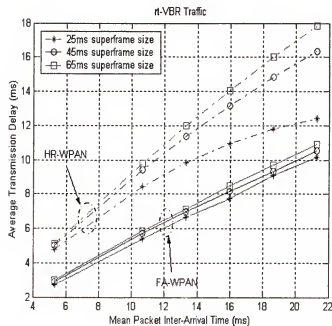


(b)

Figure 4-7. Job Failure Rate as a function of the packet inter arrival time for different superframe sizes. A) JFR of CBR traffic. B) JFR of rt-VBR traffic.



(a)



(b)

Figure 4-8. Average transmission delay as a function of the packet inter arrival time for different superframe sizes. A) Delay of CBR traffic. B) Delay of rt-VBR traffic.

The simulation results of the JFR are shown in Figure 4-7. With 25 ms superframe size, the JFR in FA-WPAN is 34% to 7% of that in HR-WPAN for the CBR traffic and is 45% to 24% of that in HR-WPAN for the rt-VBR traffic as the inter-arrival time increases. The performance differences increase with a larger superframe size. While the performance of HR-WPAN is influenced by the superframe size, the superframe size does not significantly affect the performance of FA-WPAN. Once a CTA for a DEV is allocated in a superframe in HR-WPAN, the DEV holds its transmissions for the CTA in the next superframe. Therefore, if the delay bound is shorter than the holding time, the packet will be dropped. The effect of delay bound will be described later. On the other hand, since CTAs are allocated at the packet arrival instants in FA-WPAN, more than one CTA for a DEV may be allocated in a superframe. For CBR traffic, beacon packets and E-MCTAs are more frequently generated in a short superframe than in a long one. Thus they obstruct appropriate CTA allocations. This reflects that JFR of 65 ms superframe is slightly lower than that of 25 ms superframe in Figure 4-7(a). However, this explanation is not applicable to the case of rt-VBR traffic. While the CTA location is a critical factor for the CBR traffic, fast changes of the CTA duration according to the Q-status is a critical factor for rt-VBR traffic. However, a CTA can not instantly be changed by the Q-status report. Although a DEV reports the number of pending packets to the PNC, CTAs allocated for a DEV are not changed during the current superframe and consequently non-transmitted packets in the current CTA are dropped. Thus, the CTA durations in a short superframe can be quickly adapted comparing to a long superframe. Therefore, the

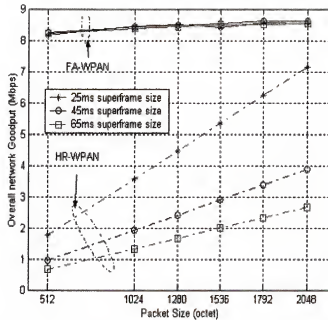


Figure 4-9. Overall network goodput as a function of the packet size

JFR of 65 ms superframe is higher than that of 25 ms superframe shown in Figure 4-7(b).

The performance differences between HR-WPAN and FA-WPAN are more obvious when the transmission delays are evaluated as shown in Figure 4-8.

Theoretically, the average transmission delay for HR-WPAN is around a half of the superframe size.

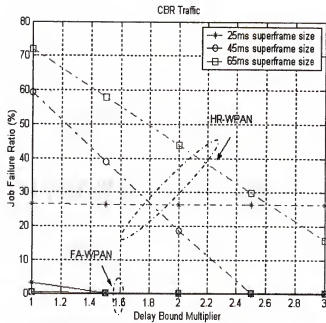
Therefore, we observe that the gradient of the performance curve for HR-WPAN with 25 ms superframe reduces to around a half of the superframe size, namely 12.5 ms.

Figure 4-9 shows the overall network goodput performance as a function of the packet size. As mentioned before, increasing packet size results in increasing packet inter-arrival time if traffic bit rates are not changed. The results in Figure 4-9 agree with the results of the JFR and transmission delays in Figure 4-7 and 4-8.

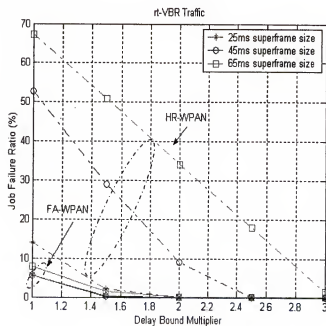
Previous evaluations are performed using the packet inter-arrival time as the delay bound. Some applications allow longer delay constraint than the inter-arrival time [40-42,

76]. Therefore, the performance using different delay bounds should be evaluated. For this purpose, we define a delay bound multiplier which is multiplier for the inter-arrival time. Here, 2048 octets packet size is used for both of traffic types. The inter-arrival times for CBR and rt-VBR traffic are 18 ms and 28 ms, respectively. Figure 4-10 shows the JFRs of both configurations. Both the JFRs of CBR and rt-VBR in FA-WPAN reaches to 0% at around 1.5 delay bound multiplier regardless of the superframe size. Although a longer delay bound generates more pending frames, the CTA duration in the 25 ms superframe for CBR traffic in HR-WPAN is insufficient to deal with more than two packets. Therefore, the JFRs for CBR in HR-WPAN are constant, regardless of the delay bounds. Except for CBR traffic case in HR-WPAN, the JFRs in HR-WPAN reach to 0% at longer delay bound multiplier than in FA-WPAN. Considering the delay bound of 30 ms, generally used maximum delay bound for MPEG-4 traffic [40, 41, 76], the higher delay bound multipliers than 1.5 may not be practical.

Figure 4-11 illustrates the JFRs of both configurations as functions of the number of flows. The 2048 octets packet size is used for both of traffic types. Delay bounds for CBR and rt-VBR traffic are three times longer than the packet inter-arrival times of both traffics because previous evaluation shows the best performance in that delay bound. For FA-WPAN, the JFRs of both traffics are constant at 0%, and then slightly increase where there are 20 flows. In the heavy load case, CTA allocation may not be synchronized with the packet arrival time because of overlapped CTAs. For HR-WPAN, the allocated CTA durations reduce with increasing number of flows so that it is not adequate to transmit all pending packets.

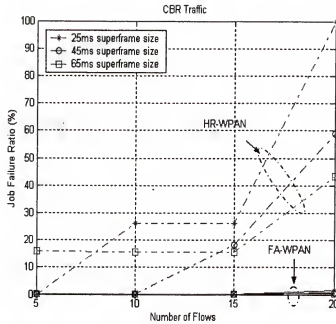


(a)

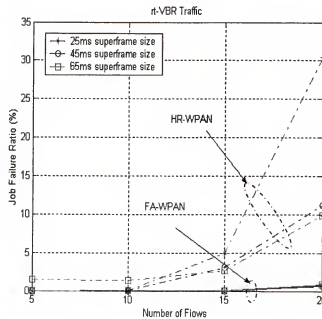


(b)

Figure 4-10. Job failure ratio as a function of the delay bound multiplier. A) JFR of CBR traffic. B) JFR of rt-VBR traffic.



(a)



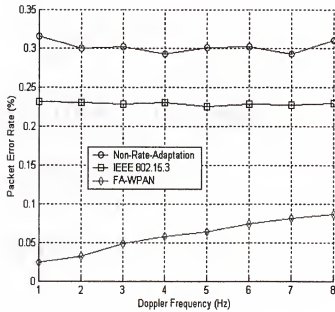
(b)

Figure 4-11. Job failure ratio as a function of the number of flows. A) JFR of CBR traffic. B) JFR of rt-VBR traffic.

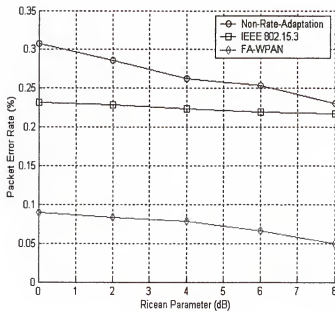
Now, the effect of a rate adaptation is evaluated in Figure 4-12. In this simulation, the superframe size is set to 35 ms and the packet size is 2048 octets. The proposed protocol is compared with two other protocols. The first one is a non-rate-adaptive protocol in which an initially chosen data rate is not changed until the flow is completed. The second one is a rate-adaptive protocol which adopts the transmission rate according to the network performance as described in IEEE 802.15.3 standard and Section 4.2.2. Under this protocol, whenever a receiver receives 10 packets, it sends the sender the received packet history such as the number of failed packets. If more than two out of the 10 packets are unsuccessfully transmitted, the sender reduces the next lower data rate. Otherwise, it increases the next higher data rate. In Figure 4-12, the first and second protocols are labeled as Non-Rate-Adaptation and IEEE 802.15.3, respectively. The time varying nature of the wireless channel is described by its Doppler spread and coherence time, which are inversely proportional to one another. In our simulation, we consider the effect of the change of the Doppler spread and coherence time with the Ricean parameter 0dB. Since WPAN is targeting on home or office environment, Doppler frequency varies from 1 Hz up to 8Hz which corresponds to the pedestrian speed of 1m/s. The coherence time is obtained [57] as

$$T_c = \frac{1}{f_m}$$

where f_m is Doppler frequency. As shown in Figure 4-12(a), the proposed scheme has the lowest PER over the other two schemes. The results illustrates that the other two schemes can hardly adapt to the varying wireless channel. Figure 4-12(b) shows the PER performance as a function of the Ricean parameter with 8Hz Doppler frequency. The PERs for all schemes reduce with increasing Ricean parameter, which means improving



(a)



(b)

Figure 4-12. Packet error rate comparisons. A) PER as functions of Doppler frequency. B) PER as a function of ricean parameter.

channel condition. We observe that the PER of FA-WPAN is up to 78 % less than that of Non-Rate-Adaptation and 76% less than that of IEEE 802.15.3.

4.5. Conclusion

In this chapter, we propose a channel time allocation algorithm assisted by feedback information. The proposed scheme targets on delay-bounded applications in HR-WPAN. The proposed algorithm initially allocates CTAs based on the statistical packet inter-arrival time, which is informed by each DEV using a modified CTRq command. The initially allocated CTAs are dynamically relocated by utilizing feedback information in order to synchronize CTA to the packet arrival time and to allocate sufficient channel time for pending packets. We verify the performance enhancement by the extensive simulations. From the simulations, we have shown that the proposed scheme gives significant performance improvements over other comparative CTA allocation schemes. We note that the performance of the proposed scheme is not influenced by variable factors such as the superframe size, a delay bound, and number of flows. The proposed method shows smaller JFR and shorter transmission delay than the HR-WPAN standard. Furthermore, a rate adaptation scheme is proposed to cope with the time varying wireless channel. In the scheme, the rate is selected according to the channel estimation results in the physical-layer. The proposed scheme reduces the packet error rate up to 89% of that of the adaptation scheme in the IEEE 802.15.3

CHAPTER 5

CONCLUSIONS AND FUTURE RESEARCH DIRECTIONS

Study of the literature has shown that designing efficient MAC protocols for wireless networks is one of the most critical aspects of network performance and proficient system operations. On the other hand, while the layering and information-hiding design paradigm is useful for managing the complexity of communication protocols, a synergy between protocol layers has to be exploited in order to achieve high performance.

In this dissertation, it is demonstrated using our proposed link-adaptive MAC protocols that such a synergistic effect can indeed deliver considerable performance improvement. Synergy is achieved through the interaction between the physical-layer and the MAC layer. Specifically, the physical-layer offers a variable throughput to the MAC layer depending on the current channel condition. Accordingly, the MAC layer allocates bandwidth by taking into account the current throughput level of the physical-layer.

In Chapter 2, we propose a link-adaptive dynamic fragmentation scheme, motivated by the realization that the physical frame size varies according to the transmission rates. In this scheme, more information bits can be transmitted over a good channel condition. Although the packet error rate of the proposed scheme causes a higher packet error rate compared with the conventional fragmentation schemes, it achieves higher throughput performance than the comparative scheme by reducing overheads.

Although the recent physical-layer specifications define multi-transmission rates and dynamic rate change based on the time-varying wireless channel condition improving

the performances of the wireless networks, the paradigms on the MAC layer for the dynamic link adaptation is not defined in the many wireless network standards. Moreover, the varying transmission rate information is the essential parameter in centralized networks with a timeslot-based or polling-based MAC protocol. Since the transmission rate decides a required channel time for the transmission of each station, it allows a central station to allocate optimal channel times to all stations.

Taking into account these considerations, enhanced MAC protocols with link adaptation capability are proposed. In Chapter 3, a link-adaptable, polling-based MAC protocol for IEEE 802.11 PCF is proposed. Utilizing the protocol, a central station can schedule an efficient polling sequence considering communication link status and number of pending frames. Consequently, higher performance gain is obtained. In Chapter 4, the MAC protocol in IEEE 802.15.3 is enhanced by proposed reporting commands. Utilizing the reported information, an allocated channel time for a communication pair can synchronize with the packet generation instant at a station, resulting in reduced transmission delays and job failure rates, which are crucial factors for time-bounded traffic. Extensive simulation studies have demonstrated that the proposed schemes produce significant performance improvement over the comparative protocols.

These encouraging results inspire us to investigate an extension of the link adaptation concept to not only the other parameters, but also higher protocol layers such as routing protocols for wireless communication. In this dissertation, we demonstrate the effectiveness of MAC protocol-assisted rate adaptation, which is one of the link adaptation techniques. The other notable mechanism using link adaptation is power control. In an emerging sensor network in which power conservation is a crucial factor,

the transmission power can be conserved according to the link condition. Furthermore, in wireless networks using a multi-carrier system or directional antenna, power can be allocated to each carrier or in different directions based on its link condition to meet performance criteria. Since the transmission time and the power consumed vary according to the channel condition, this information helps to enhance the performance of a routing protocol in multi-hop networks.

Our future research direction will be to investigate a novel mechanism using the link-adaptable cross-layer design for performance improvement over wireless networks.

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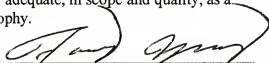
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BIOGRAPHICAL SKETCH

Byung-Seo Kim began his college career as an electrical engineering student at In-Ha University in InChon, Korea. He received the B.S. degree in electrical engineering from In-Ha University in 1998. Between 1997 and 1999, he worked for Motorola Korea Ltd., PaJu-city, Korea, as a Computer Integrated Manufacturing (CIM) Engineer. He received his M.S. degree in electrical and computer engineering from the University of Florida (UF), Gainesville, Florida, in 2001. He received the Ph.D. degree in electrical and computer engineering from UF in December 2004. His main research area is designing and developing efficient link-adaptable medium access control protocols over high-speed wireless networks. These areas include cross-layered wireless protocol design; QoS guarantees for wireless network protocols designs for multimedia applications; resource allocation algorithm development for multi-carrier communication systems such as OFDM; fair scheduling for high-speed wireless networks; OFDM Implementation and optimization with MAC layer in high-speed Wireless LANs; physical layer design and implementation using OFDM; physical layer design for powerline communication; medium access control for wireless sensor networks. In addition to the above, Byung-Seo Kim has extensive research experience in the areas of computer network design and cellular data networks.

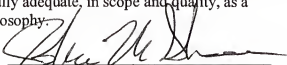
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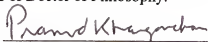

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